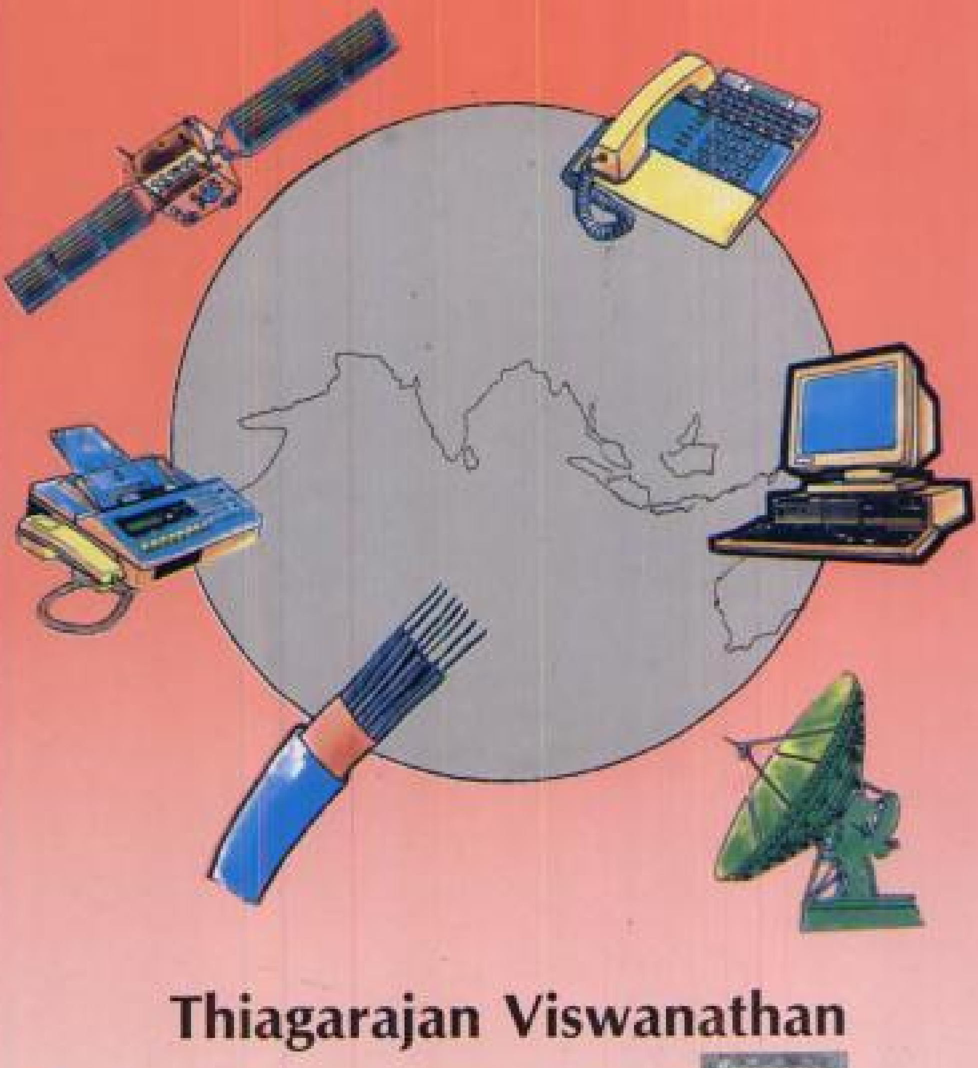
Eastern Economy Edition

Telecommunication Switching Systems and Networks



Thiagarajan Viswanathan

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TELECOMMUNICATION SWITCHING SYSTEMS AND NETWORKS
by Thiagarajan Viswanathan

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Contents

Foreword

For a number of years, it has become the fashion to write books on analytic themes rather than on topics pertaining to practical systems and their synthesis. That has been so mainly for two reasons: Firstly, analytic themes lend themselves to elegant pedagogic presentation while engineering practices do not. Secondly, practical systems change rapidly, and become dated pretty fast while analytical theory remains valid for long periods of time. In any case, there are few books available on the current practice of telecommunication systems. That leads to a vicious circle — in the absence of books, the topic is not taught in universities, and as it is not taught in universities, books are not written. Thiagarajan Viswanathan has written a book which breaks this vicious circle, and makes a laudable attempt to fill a major gap.

In the next twenty years, we may expect to witness revolutionary changes in telecommunications practice. The foundations for such developments have already been laid in the form of ISDN. Hence, a book on telecommunications systems based on the newly accepted international practices is timely.

In the flurry and excitement of new developments, the tendency is to forget the pioneering past, and thereby lose the historical perspective so essential for scholarly study. I am, therefore, particularly pleased that his book does pay attention to the historical processes in telecommunication switching.

I am happy to commend this book to all telecommunication engineers.

P.V. Indiresan
President
The Institution of Electronics and
Telecommunication Engineers
New Delhi

Preface

Today's telecommunication network is a complex interconnection of a variety of heterogeneous switching systems. Electromechanical and electronic systems, direct and common control systems, and hard-wired and stored program control systems coexist. In a sense, it is a marvel that these systems work in close cooperation to offer a plethora of complex telecommunication services, often involving instantaneous information transfer across the globe. Presently, two important classes of telecommunication networks, viz. public switched telephone network (PSTN) and public data network (PDN) are in wide use. The newly emerging integrated services digital network (ISDN) is expected to be in place in the next 20 years or so as a result of the process of total digitalisation of telecommunication networks currently under way. This text is a treatment on both switching systems and telecommunication networks in a single volume.

The motivation for writing this text came when I taught regular full-semester and short-term courses on 'switching systems and networks' at the Indian Institute of Science, Bangalore. I keenly felt the absence of a suitable text for the purpose. This book is meant to fill this void and is designed for the final year undergraduate or the first year postgraduate students in electronics and communications engineering and allied subjects. It may be difficult to cover the entire text in one semester. Depending on other courses offered and the emphasis given in a programme, a teacher may like to omit one or two chapters in a one-semester course.

I have attempted to give a balanced blend of theoretical and practical aspects in the text. Concepts and system level treatment are given emphasis. Analytical or mathematical treatment is introduced only to the extent required. Worked-out examples are given where considered necessary. All chapters contain exercises, and answers are provided for the selected exercises at the end of the book.

For over 40 years, telecommunications has largely been confined to the private domain of network operators. Research, development and even education have been pursued by a few firms and organisations. It is only recently that a large number of entrepreneurs have entered the field of telecommunications. Such new entrants should find this book to be a valuable asset. The coverage of recent topics like fibre optic communication systems and networks, time division switching systems, data networks, ISDN, and voice data integration schemes should interest the practising professionals.

I have devoted two full chapters to discuss at length, the somewhat outdated Strowger and crossbar systems, for two reasons. The first and most important one is pedagogical. Many fundamental concepts underlying the design of

modern electronic exchanges have evolved from these systems. Secondly, most of the less developed and developing countries including India have operational Strowger and crossbar systems, often in large numbers.

Chapter 1 introduces the subject. In this chapter, the evolution of the telecommunication networks is briefly traced, starting from the invention of the telephone by Alexander Graham Bell and ending with the emerging ISDN. A classification scheme for the switching systems is presented. Basic network structures such as folded, nonfolded, blocking and nonblocking structures are introduced.

Chapter 2 deals with pulse dialling and Strowger automatic switching systems. A set of parameters to evaluate alternative designs of switching systems is introduced in this chapter. These parameters are generic in nature and are used throughout the text to compare different designs.

Chapter 3 discusses the dual tone multifrequency (DTMF) telephones and signalling, the common control concepts, and the crossbar switching systems.

Chapter 4 is devoted to stored program control (SPC) and multistage space division networks. Here, fault tolerant SPC architectures are discussed besides system and application software aspects. The enhanced telecommunication services that become possible with the introduction of SPC are then presented.

Chapter 5 lays the foundation for digital voice transmission. After covering linear quantisation, companding and CCITT *A*-law are discussed. This chapter ends with a presentation on CCITT time division multiplexing hierarchy.

Chapter 6 concentrates on time division switching. First, analog and digital time division switching techniques are discussed. The idea of time multiplexed input/output streams and the corresponding time division switching concepts are then presented. At the end, time-space combination configurations are discussed with real life examples.

Chapter 7 is devoted to fibre optic communication systems which are emerging as a major alternative to coaxial cable systems. This chapter covers types of optical fibres, optical sources and detectors, and deals with power losses in fibre optic systems giving related power budget calculations. This chapter concludes with a discussion on the practical application of fibre optic communication systems in telecommunication networks.

Chapter 8 is on traffic engineering which is the basis for the design and analysis of telecommunication networks. Grade of service (GOS) and blocking probability ideas are placed in proper perspective in this chapter. Basic concepts of modelling switching systems as birth-death stochastic processes are presented. Loss system and delay system models are discussed.

Chapters 9–11 deal with the three most important telecommunication networks: telephone networks, data networks and integrated services digital networks. Chapter 9 provides a comprehensive coverage of the telephone network aspects discussing subscriber loop systems, switching hierarchy, and transmission, numbering and charging plans. In addition, a brief description of

the various transmission systems, viz. coaxial cable, ionospheric, troposcatter, microwave, and satellite communication systems, is given. Besides, a discussion on inchannel and common channel signalling systems is also included. Finally, this chapter presents the introductory concepts of the newly emerging cellular mobile communications.

Chapter 10 opens with a discussion on data transmission over PSTN and provides a detailed treament on open system interconnection (OSI) reference model. It then goes on to present important aspects of local and metropolitan area networks, and satellite based data networks. Basics of fibre optic data networks where considerable research interest lies at present are then dealt with. Other aspects discussed in this chapter include data network standards and internetworking.

In Chapter 11, after briefly discussing the motivation for ISDN, some of the new services that are possible in the context of ISDN are presented. ISDN architecture, user-network interface and ISDN standards are covered in this chapter. It is envisioned that artificial intelligence and expert systems would play a significant role in future telecommunication networks and hence a brief treatise on this topic is given. The chapter concludes with a discussion on some of the voice data integration schemes.

I set out to write this text with an aim of giving a broad, yet fairly in-depth, and up-to-date coverage of telecommunication switching systems and networks. How far I have succeeded in this aim is for the readers to judge. I would be grateful for comments from the readers, especially students, teachers and practising professionals.

T. Viswanathan

Acknowledgements

Many have contributed to the successful preparation of this text. I would like to place on record my grateful thanks to each one of them.

I began writing this text when I was a Professor at the Indian Institute of Science (IISc), Bangalore, but wrote a major part of it while being the Director of the Indian National Scientific Documentation Centre (INSDOC), a constituent establishment of the Council of Scientific and Industrial Research (CSIR) of the Government of India. Financial support for the preparation of the manuscript came from the Curriculum Development Cell at IISc set up by the Ministry of Human Resource Development. The excellent infrastructural facilities of INSDOC and the gracious words of encouragement from Prof. S K Joshi, Director General, CSIR hastened the process of completing the text.

Shri J Gopal of the Department of Telecommunications put in considerable effort and reviewed the manuscript in a time-bound manner. Shri J M Jose, a research fellow at INSDOC, verified the worked-out examples and meticulously perused parts of the manuscript. He also rendered very valuable assistance in many other ways.

Smt. Chandrika Sridhar at IISc and Smt. Sushma Arora at INSDOC rendered their skillful services in word processing the handwritten manuscript. Both of them did their job so efficiently and delightfully that I had no hesitation in revising, modifying and correcting the computerised manuscript many times. Both of them went out of their way to meet deadlines and schedules.

The camera ready copies of the manuscript were prepared at INSDOC using desk top publishing facilities. Shri B Sadananda Rao, Smt. Sarla Dutta and Shri S D Barman contributed significantly to this activity. The rich experience, expertise and the maturity of Shri B Sadananda Rao have been an asset.

All my office staff and a few other colleagues at INSDOC have in some way contributed to this process. In particular, S/Shri P R Gupta, Trilok Singh Negi, Durga Dutt Tiwari and Balwant Singh deserve mention.

The publishers, Prentice-Hall of India, meticulously processed the manuscript with remarkable speed, both during the editorial and production stages, and made valuable improvements.

A number of persons have been well wishers of this activity. Foremost among them are S/Shri N Jayaraman, V Rajaraman, T N Seshan, Dr B B Sunderesan, Y S Rajan and N Pant.

My wife has been a mentor in this effort. When I was overjoyed at having completed some portion, she gently reminded me of the work still left. When I was concerned about not progressing, she took care of even my trivial personal needs, enabling me to devote my full energy on this effort. The text was tried on

my first daughter, a bright student of mathematics, who studied the text and learnt the subject on her own. When she did not understand some concepts, it was an indication for me to revise the concerned portion. When I was excited about some new activities and talked of some 'big things', it was given to my younger daughter to say 'Appa, first finish your book, that is the best service you can render'.

I am overwhelmed, when I think of the fact that there are so many who have worked to make this effort a success although there is only one name printed as author in the text. I am indebted to each one of them.

It is my experience that both science and religion have their roles to play in one's life. While science has helped me to think and reason rationally, religion has carried me beyond the realm of thought and reasoning. A great seer of India has blessed this effort and I feel that he has taken me one step nearer to God through this effort. It is with great humility that I offer this text at the feet of the Supreme Being.

T. Viswanathan

1

Introduction

The field of telecommunications has evolved from a stage when signs, drum beats and semaphores were used for long distance communication to a stage when electrical, radio and electro-optical signals are being used. Optical signals produced by laser sources and carried by ultra-pure glass fibres are recent additions to the field. Telecommunication networks carry information signals among entities which are geographically far apart. An entity may be a computer, a human being, a facsimile machine, a teleprinter, a data terminal, and so on. Billions of such entities the world-over are involved in the process of information transfer which may be in the form of a telephone conversation or a file transfer between two computers or a message transfer between two terminals, etc. In telephone conversation, the one who initiates the call is referred to as the **calling subscriber** and the one for whom the call is destined is the **called subscriber**. In other cases of information transfer, the communicating entities are known as **source** and **destination**, respectively.

The full potential of telecommunications is realised only when any entity in one part of the world can communicate with any other entity in another part of the world. Modern telecommunication networks attempt to make this idea of 'universal connectivity' a reality. Connectivity in telecommunication networks is achieved by the use of switching systems. This text deals with the telecommunication switching systems and the networks that use them to provide worldwide connectivity.

1.1 Evolution of Telecommunications

Historically, transmission of telegraphic signals over wires was the first technological development in the field of modern telecommunications. Telegraphy was introduced in 1837 in Great Britain and in 1845 in France. In March 1876, Alexander Graham Bell demonstrated his telephone set and the possibility of telephony, i.e. long-distance voice transmission. Graham Bell's invention was one of those rare inventions which was put to practical use almost immediately. His demonstrations laid the foundation for telephony.

Graham Bell demonstrated a point-to-point telephone connection. A network using point-to-point connections is shown in Fig. 1.1. In such a

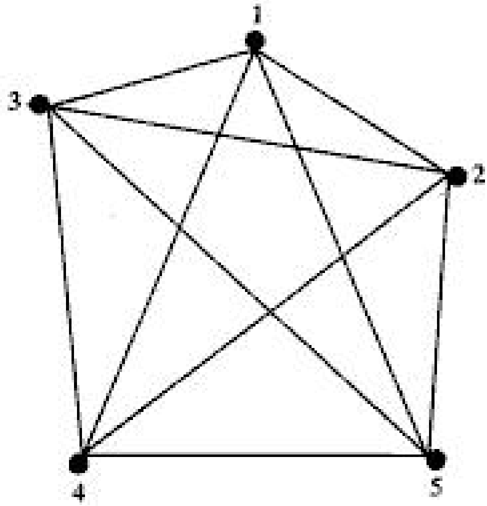


Fig. 1.1 A network with point-to-point links.

network, a calling subscriber chooses the appropriate link to establish connection with the called subscriber. In order to draw the attention of the called subscriber before information exchange can begin, some form of **signalling** is required with each link. If the called subscriber is engaged, a suitable indication should be given to the calling subscriber by means of signalling.

In Fig. 1.1, there are five entities and 10 point-to-point links. In a general case with n entities, there are $n(n-1)/2$ links. Let us consider the n entities in some order. In order to connect the first entity to all other entities, we require $(n-1)$ links. With this, the second entity is already connected to the first. We now need $(n-2)$ links to connect the second entity to the others. For the third entity, we need $(n-3)$ links, for the fourth $(n-4)$ links, and so on. The total number of links, L, works out as follows:

$$L = (n-1) + (n-2) + \ldots + 1 + 0 = n(n-1)/2 \tag{1.1}$$

Networks with point-to-point links among all the entities are known as **fully connected networks**. The number of links required in a fully connected network becomes very large even with moderate values of n. For example, we require 1225 links for fully interconnecting 50 subscribers. Consequently, practical use of Bell's invention on a large scale or even on a moderate scale demanded not only the telephone sets and the pairs of wires, but also the so-called **switching system** or the **switching office** or the **exchange**. With the introduction of the switching systems, the subscribers are not connected directly to one another; instead, they are connected to the switching system as shown in Fig. 1.2. When a subscriber wants to communicate with another,

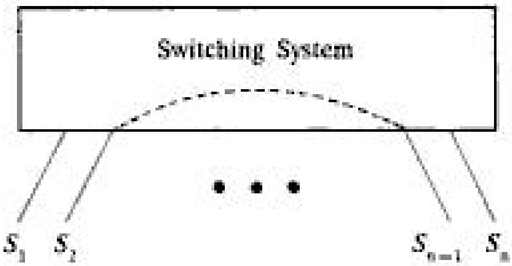


Fig. 1.2 Subscriber interconnection using a switching system.

a connection is established between the two at the switching system. Figure 1.2 shows a connection between subscriber S_2 and S_{n-1}. In this configuration, only one link per subscriber is required between the subscriber and the switching system, and the total number of such links is equal to the number of subscribers connected to the system. Signalling is now required to draw the attention of the switching system to establish or release a connection. It should also enable the switching system to detect whether a called subscriber is busy and, if so, indicate the same to the calling subscriber. The functions performed by a switching system in establishing and releasing connections are known as **control functions**.

Early switching systems were manual and operator oriented. Limitations of operator manned switching systems were quickly recognised and automatic exchanges came into existence. Automatic switching systems can be classified as **electromechanical** and **electronic**. Electromechanical switching systems include **step-by-step** and **crossbar systems**. The step-by-step system is better known as Strowger switching system after its inventor A.B. Strowger. The control functions in a Strowger system are performed by circuits associated with the switching elements in the system. Crossbar systems have hard-wired control subsystems which use relays and latches. These subsystems have limited capability and it is virtually impossible to modify them to provide additional functionalities. In electronic switching systems, the control functions are performed by a computer or a processor. Hence, these systems are called **stored program control (SPC)** systems. New facilities can be added to a SPC system by changing the control program. The switching scheme used by electronic switching systems may be either **space division switching** or **time division switching**. In space division switching, a dedicated path is established between the calling and the called subscribers for the entire duration of the call. Space division switching is also the technique used in Strowger and crossbar systems. An electronic exchange may use a crossbar switching matrix for space division switching. In other words, a crossbar switching system with SPC qualifies as an electronic exchange.

In time division switching, sampled values of speech signals are transferred at fixed intervals. Time division switching may be analog or digital. In analog switching, the sampled voltage levels are transmitted as they are,

whereas in digital switching, they are binary coded and transmitted. If the coded values are transferred during the same time interval from input to output, the technique is called **space switching**. If the values are stored and transferred to the output at a later time interval, the technique is called **time switching**. A time division digital switch may also be designed by using a combination of space and time switching techniques. Figure 1.3 summarises

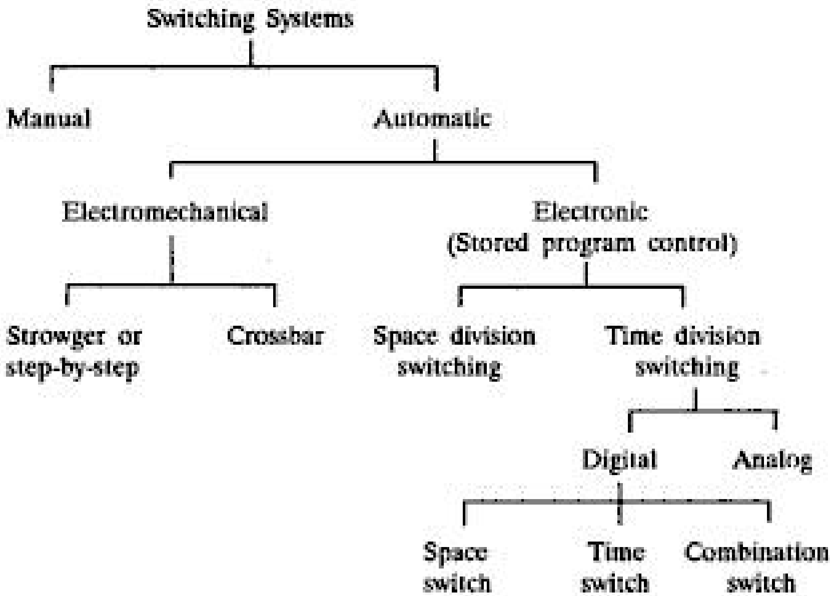


Fig. 1.3 Classification of switching systems.

the classification of switching systems. In Chapters 2 and 3, we deal with electromechanical switching systems. Electronic space division networks are discussed in Chapter 4. Digitisation of speech, which is a fundamental requirement for electronic time division switching networks, is discussed in Chapter 5, and the time division switching techniques in Chapter 6.

Subscribers all over the world cannot be connected to a single switching system unless we have a gigantic switching system in the sky and every subscriber has a direct access to the same. Although communication satellite systems covering the entire globe and low cost roof-top antenna present such a scenario, the capacity of such systems is limited at present. The major part of the telecommunication networks is still ground based, where subscribers are connected to the switching system via copper wires. Technological and engineering constraints of signal transfer on a pair of wires necessitate that the subscribers be located within a few kilometres from the switching system. By introducing a number of stand-alone switching systems in appropriate geographical locations, communication capability can be established among

the subscribers in the same locality. However, for subscribers in different localities to communicate, it is necessary that the switching systems are interconnected in the form of a network. Figure 1.4 shows a telecommunication network. The links that run between the switching systems are called **trunks,** and those that run to the subscriber premises are known as **subscriber lines.** The number of trunks may vary between pairs of switching systems and is determined on the basis of traffic between them. As the number of switching systems increases, interconnecting them becomes complex. The problem is tackled by introducing a hierarchical structure among the switching systems and using a number of them in series to establish connection between subscribers. The design and analysis of switching systems and telecommunication networks are based on the traffic engineering concepts; these are covered in Chapter 8.

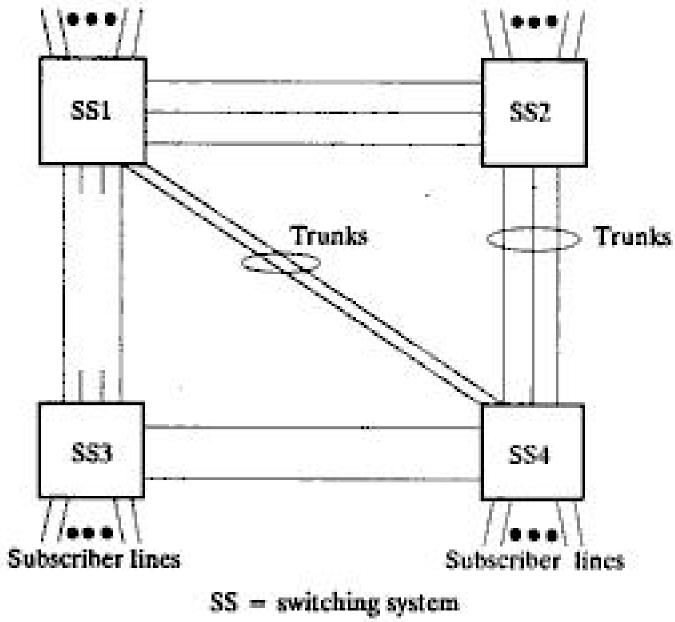
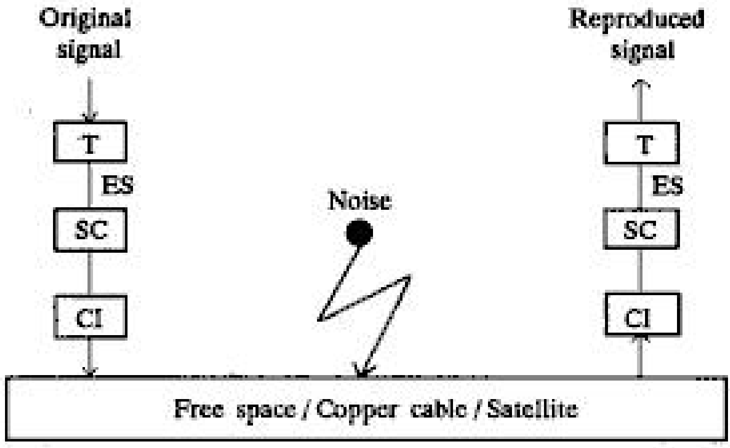
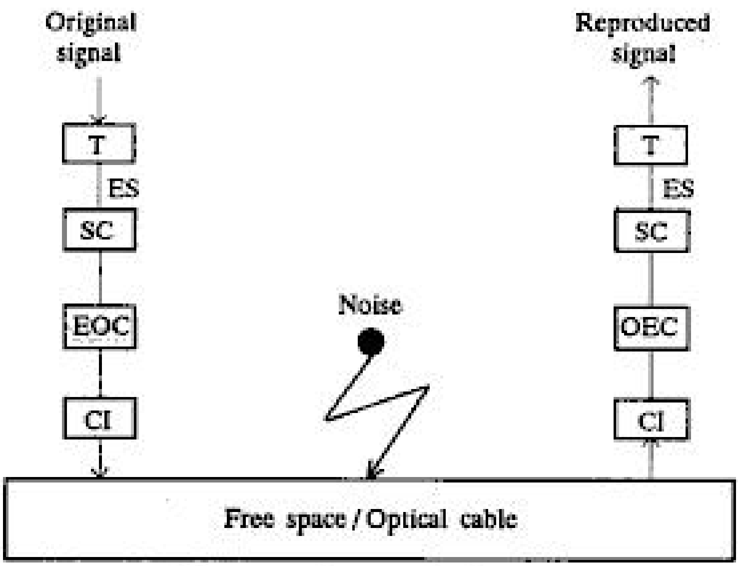


Fig. 1.4 A telecommunication network.

A modern telecommunication network may be viewed as an aggregate of a large number of point-to-point electrical or optical communication systems shown in Fig. 1.5. While these systems are capable of carrying electrical or optical signals, as the case may be, the information to be conveyed is not always in the form of these signals. For example, human speech signals need to be converted to electrical or optical signals before they can be carried by a communication system. Transducers perform this energy conversion. Present day optical sources require electrical signals as input, and the optical



(a) An electrical communication system



(b) An optical communication system

CI = channel interface EOC = electrical to optical converter
ES = electrical signal OEC = optical to electrical converter
SC = signal conditioner T = transducer.

Fig. 1.5 Elements of a communication system.

detectors produce electrical signals as output. Hence, the original signals are first converted to electrical signals and then to optical signals at the transmitting end of an optical communication system and at the receiving end optical signals are converted to electrical signals before the original signal is reproduced. A medium is required to carry the signals. This medium, called the channel, may be the free space, a copper cable, or the free space in conjunction with a satellite in the case of an electrical communication system. An optical communication system may use the line-of-sight free space or fibre optic cables as the channel. Channels, in general, are lossy and prone to external noise that corrupts the information carrying signals. Different channels exhibit different loss characteristics and are affected to different degrees by noise. Accordingly, the chosen channel demands that the information signals be properly conditioned before they are transmitted, so that the effect of the lossy nature of the channel and the noise is kept within limits and the signals reach the destination with acceptable level of intelligibility and fidelity. Signal conditioning may include amplification, filtering, band-limiting, multiplexing and demultiplexing. Fibre optic communication systems are emerging as major transmission systems for telecommunications. Chapter 7 deals with this newly emerging communication system.

The channel and the signal characteristics of individual communication systems in a telecommunication network may vary widely. For example, the communication system between the subscriber and the switching system uses most often a pair of copper wires as the channel, whereas the communication system between the switching systems may use a coaxial cable or the free space (microwave) as the channel. Similarly, the type of end equipment used at the subscriber premises would decide the electrical characteristics of signals carried between the subscriber end and the switching system. For example, electrical characteristics of teleprinter signals are completely different from those of telephone signals. In fact, such wide variations in signal characteristics have led to the development of different service specific telecommunication networks that operate independently. Examples are:

1. Telegraph networks
2. Telex networks
3. Telephone networks
4. Data networks.

We discuss the telephone networks in Chapter 9 and the data networks in Chapter 10. Management and maintenance of multiple networks are expensive. The question then arises: Is it possible to design a single network that can carry all the services? The key to the solution of this problem lies in the digitalisation of services. If all the service specific signals can be converted to a common digital domain, a network capable of transporting digital signals can carry the multitude of services. This approach is leading to

the evolution of the integrated services digital network (ISDN) which is discussed in Chapter 11.

1.2 Simple Telephone Communication

In the simplest form of a telephone circuit, there is a one way communication involving two entities, one receiving (listening) and the other transmitting (talking). This form of one way communication shown in Fig. 1.6 is known as

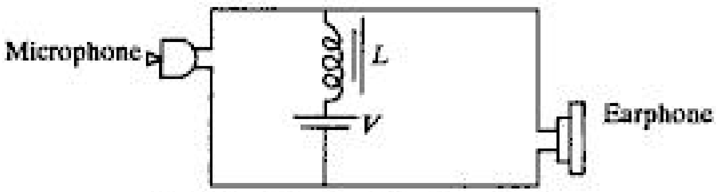


Fig. 1.6 A simplex telephone circuit.

simplex communication. The microphone and the earphone are the transducer elements of the telephone communication system. Microphone converts speech signals into electrical signals and the earphone converts electrical signals into audio signals. Most commonly used microphone is a carbon microphone. Carbon microphones do not produce high fidelity signals, but give out strong electrical signals at acceptable quality levels for telephone conversation. In carbon microphones, a certain quantity of small carbon granules is placed in a box. Carbon granules conduct electricity and the resistance offered by them is dependent upon the density with which they are packed. One side of the box cover is flexible and is mechanically attached to a diaphragm. When sound waves impinge on the diaphragm, it vibrates, causing the carbon granules to compress or expand, thus changing the resistivity offered by the granules. If a voltage is applied to the microphone, the current in the circuit varies according to the vibrations of the diaphragm. The theory of the carbon microphone indicates that the microphone functions like an amplitude modulator. When the sound waves impinge on the diaphragm, the instantaneous resistance of the microphone is given by

$$r_i = r_0 - r \sin \omega t \qquad (1.2)$$

where

r_0 = quiescent resistance of the microphone when there is no speech signal

r = maximum variation in resistance offered by the carbon granules, $r < r_0$

r_i = instantaneous resistance.

The negative sign in Eq. (1.2) indicates that when the carbon granules are compressed the resistance decreases and vice versa. Ignoring impedances

external to the microphone in the circuit given in Fig. 1.6, without loss of generality, the instantaneous current in the microphone is given by

$$i = V/(r_0 - r \sin \omega t) = I_0(1 - m \sin \omega t)^{-1} \tag{1.3}$$

where

$I_0 = V/r_0$ = quiescent current in the microphone

$m = r/r_0, \quad m < 1$

By binomial theorem, Eq. (1.3) may be expanded as

$$i = I_0(1 + m \sin \omega t + m^2 \sin^2 \omega t + \ldots) \tag{1.4}$$

If the value of m is sufficiently small, which is usually the case in practice, higher-order terms can be ignored in Eq. (1.4), giving thereby

$$i = I_0(1 + m \sin \omega t) \tag{1.5}$$

which resembles the amplitude modulation (AM) equation. Thus, the carbon granule microphone acts as a modulator of the direct current I_0 which is analogous to the carrier wave in AM systems. The quantity m is equivalent to the modulation index in AM. The higher-order terms in Eq. (1.4) represent higher-order harmonic distortions, and hence it is essential that the value of m be kept sufficiently low. In Eq. (1.5), the alternating current output i is zero if the quiescent current I_0 is zero. Hence, the flow of this steady current through the microphone is essential, and the current itself is known as energising current. In Fig.1.6, the inductor acts as a high impedance element for voice frequency signals but permits the d.c. from the battery to flow to the microphone and the receiver. The voice frequency signals generated by the microphone reach the earphone without being shunted by the battery arm and are converted to audio signals there.

The earphone is usually an electromagnet with a magnetic diaphragm which is positioned such that there is an air gap between it and the poles of the electromagnet. When the electromagnet is energised by passing a current, a force is exerted on the diaphragm. The voice frequency current from the microphone causes variation in the force exerted by the electromagnet, thus vibrating the diaphragm and producing sound waves. Faithful reproduction of the signals by the receiver requires that the magnetic diaphragm be displaced in one direction from its unstressed position. The quiescent current provides this bias. In some circuits, a permanent magnet is used to provide the necessary displacement instead of the quiescent current. The instantaneous flux linking the poles of the electromagnet and the diaphragm is given by

$$\phi_i = \phi_0 + \phi \sin \omega t \tag{1.6}$$

where

ϕ_0 = constant flux due to the quiescent current or the permanent magnet

ϕ = maximum amplitude of flux variation, $\phi < \phi_0$

ϕ_i = instantaneous flux

Equation (1.6) assumes that the vibrations of the diaphragm are too small to affect the length of the air gap and that the reluctance of the magnetic path is constant. The instantaneous force exerted on the diaphragm is proportional to the square of the instantaneous flux linking the path. Therefore,

$$F = K(\phi_0 + \phi \sin \omega t)^2 \tag{1.7}$$

where K is the constant of proportionality. Expanding the right-hand side of Eq. (1.7), we have

$$F = K(\phi_0^2 + \phi^2 \sin^2 \omega t + 2\phi_0\phi \sin \omega t) \tag{1.8}$$

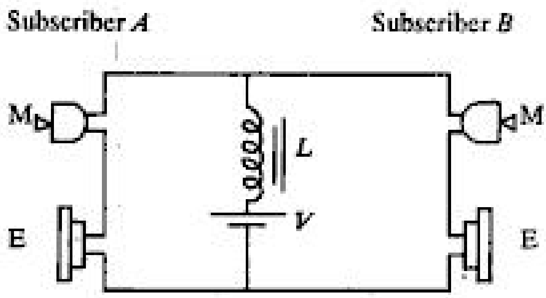
When $(\phi / \phi_0) \ll 1$, we can ignore the second-order term in Eq.(1.8). We then have

$$F = K\phi_0^2(1 + K_1 I_0 \sin \omega t) \tag{1.9}$$

where $I_0 \sin \omega t$ is the current flowing through the coil. We thus see that the force experienced by the diaphragm is in accordance with the signals produced by the microphone.

In a normal telephone communication system, information is transferred both ways. An entity is capable of both receiving and sending although these do not take place simultaneously. An entity is either receiving or sending at any instant of time. When one entity is transmitting, the other is receiving and vice versa. Such a form of communication where the information transfer takes place both ways but not simultaneously is known as **half-duplex communication**. If the information transfer takes place in both directions simultaneously, then it is called **full-duplex communication.**

Figure 1.6 may be modified to achieve half-duplex communication by the introduction of a transmitter and receiver at both ends of the circuit as shown in Fig. 1.7. In this circuit, the speech of *A* is heard by *B* as well as in *A*'s own earphone. This audio signal, heard at the generating end, is called **sidetone**. A certain amount of sidetone is useful, or even essential. Human speech and hearing system is a feedback system in which the volume of speech is automatically adjusted, based on the sidetone heard by the ear. If no sidetone is present, a person tends to shout, and if too much of sidetone is present, there is a tendency to reduce the speech to a very low level. In the circuit of Fig. 1.7, the entire speech intensity is heard as sidetone, which is not desirable. Figure 1.8 gives a circuit where a small level of sidetone and the



E = earphone M = microphone

Fig. 1.7 A half-duplex telephone circuit.

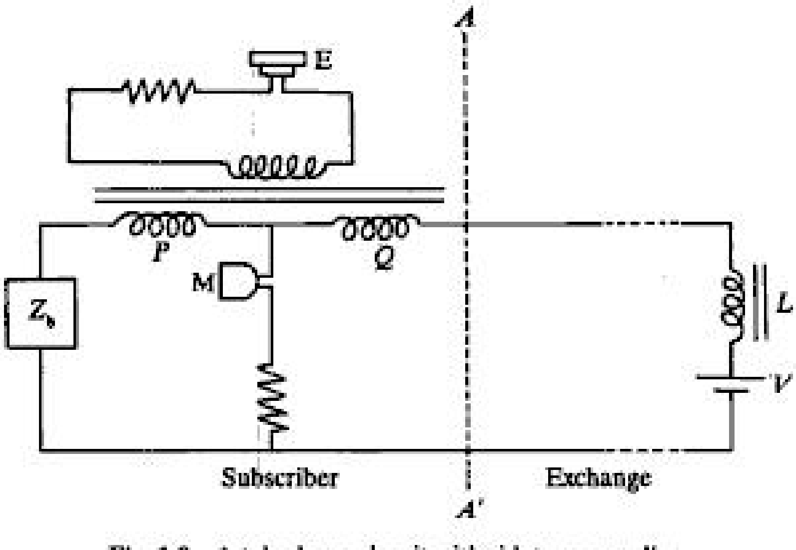


Fig. 1.8 A telephone circuit with sidetone coupling.

full speech signal from the other party are coupled to the receiver. The impedance Z_b is chosen to be more or less equal to the impedance seen by the circuit to the right of section AA'. As a consequence, with proper side-tone coupling the speech signal from the microphone M divides more or less equally in the two windings P and Q. Since the signals in these two windings are in the opposite direction, only a small induced voltage appears in the receiver circuit providing the sidetone. When a signal is received from the other entity, it travels in the same direction in both windings P and Q, inducing a large signal in the receiver circuit.

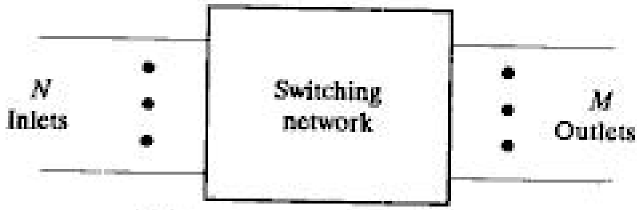
1.3 Basics of a Switching System

A major component of a switching system or an exchange is the set of input and output circuits called **inlets** and **outlets**, respectively. The primary function of a switching system is to establish an electrical path between a given inlet-outlet pair. The hardware used for establishing such a connection is called the **switching matrix** or the **switching network**. It is important to note that the switching network is a component of the switching system and should not be confused with telecommunication network. Figure 1.9(a) shows a model of a switching network with N inlets and M outlets. When $N = M$, the switching network is called a **symmetric network**. The inlets/outlets may be connected to local subscriber lines or to trunks from/to other exchanges as shown in Fig. 1.9(b). When all the inlets/outlets are connected to the subscriber lines, the logical connection appears as shown in Fig. 1.9(c). In this case, the output lines are folded back to the input and hence the network is called a **folded network**. In Fig. 1.9(b), four types of connections may be established:

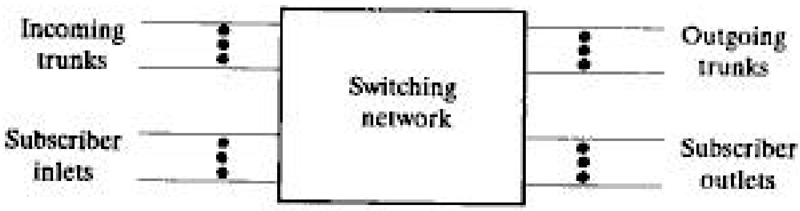
1. Local call connection between two subscribers in the system
2. Outgoing call connection between a subscriber and an outgoing trunk
3. Incoming call connection between an incoming trunk and a local subscriber
4. Transit call connection between an incoming trunk and an outgoing trunk.

In a folded network with N subscribers, there can be a maximum of $N/2$ simultaneous calls or information interchanges. The switching network may be designed to provide $N/2$ simultaneous switching paths, in which case the network is said to be **nonblocking**. In a nonblocking network, as long as a called subscriber is free, a calling subscriber will always be able to establish a connection to the called subscriber. In other words, a subscriber will not be denied a connection for want of switching resources. But, in general, it rarely happens that all the possible conversations take place simultaneously. It may, hence, be economical to design a switching network that has as many simultaneous switching paths as the average number of conversations expected. In this case, it may occasionally happen that when a subscriber requests a connection, there are no switching paths free in the network, and hence he is denied connection. In such an event, the subscriber is said to be blocked, and the switching network is called a **blocking network**. In a blocking network, the number of simultaneous switching paths is less than the maximum number of simultaneous conversations that can take place. The probability that a user may get blocked is called **blocking probability**.

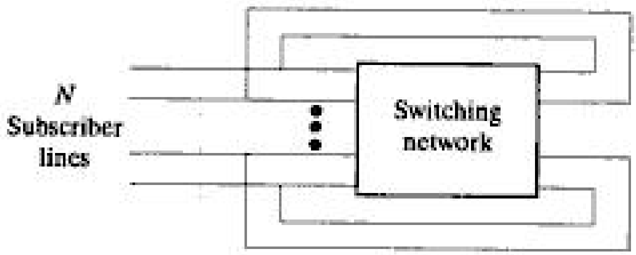
All the switching exchanges are designed to meet an estimated maximum average simultaneous traffic, usually known as **busy hour traffic**. Past records of the telephone traffic indicate that even in a busy exchange, not



(a) Model of a switching network



(b) Inlets/outlets connections



(c) Folded network

(d) Nonfolded network

Fig. 1.9 Switching network configurations.

more than 20–30 per cent of the subscribers are active at the same time. Hence, switching systems are designed such that all the resources in a system are treated as common resources and the required resources are allocated to a conversation as long as it lasts. The quantum of common resources is determined based on the estimated busy hour traffic. When the traffic exceeds the limit to which the switching system is designed, a subscriber experiences blocking. A good design generally ensures a low blocking probability.

The traffic in a telecommunication network is measured by an internationally accepted unit of traffic intensity known as **erlang** (E), named after an illustrious early contributor to traffic theory. A switching resource is said to carry one erlang of traffic if it is continuously occupied throughout a given period of observation. Teletraffic concepts are discussed in Chapter 8.

In a switching network, all the inlet/outlet connections may be used for interexchange transmission. In such a case, the exchange does not support local subscribers and is called a **transit exchange**. A switching network of this kind is shown in Fig. 1.9(d) and is called a **nonfolded network**. In a nonfolded network with N inlets and N outlets, N simultaneous information transfers are possible. Consequently, for a nonfolded network to be nonblocking, the network should support N simultaneous switching paths.

While the switching network provides the switching paths, it is the control subsystem of the switching system that actually establishes the path. The switching network does not distinguish between inlets/outlets that are connected to the subscribers or to the trunks. It is the job of the control subsystem to distinguish between these lines and interpret correctly the signalling information received on these lines. It senses the end of information transfer and releases connections. A connection is established, based on the signalling information received on the inlet lines. The control subsystem sends out signalling information to the subscriber and other exchanges connected to the outgoing trunks. In addition, signalling is also involved between different subsystems within an exchange. The signalling formats and requirements for the subscriber, the trunks and the subsystems differ significantly. Accordingly, a switching system provides for three different forms of signalling:

1. Subscriber loop signalling
2. Interexchange signalling
3. Intraexchange or register signalling.

A switching system is composed of elements that perform switching, control and signalling functions. Figure 1.10 shows the different elements of a switching system and their logical interconnections. The subscriber lines are terminated at the subscriber line interface circuits, and trunks at the trunk interface circuits. There are some service lines used for maintenance and testing purposes. Junctor circuits imply a folded connection for the local subscribers and the service circuits. It is possible that some switching systems

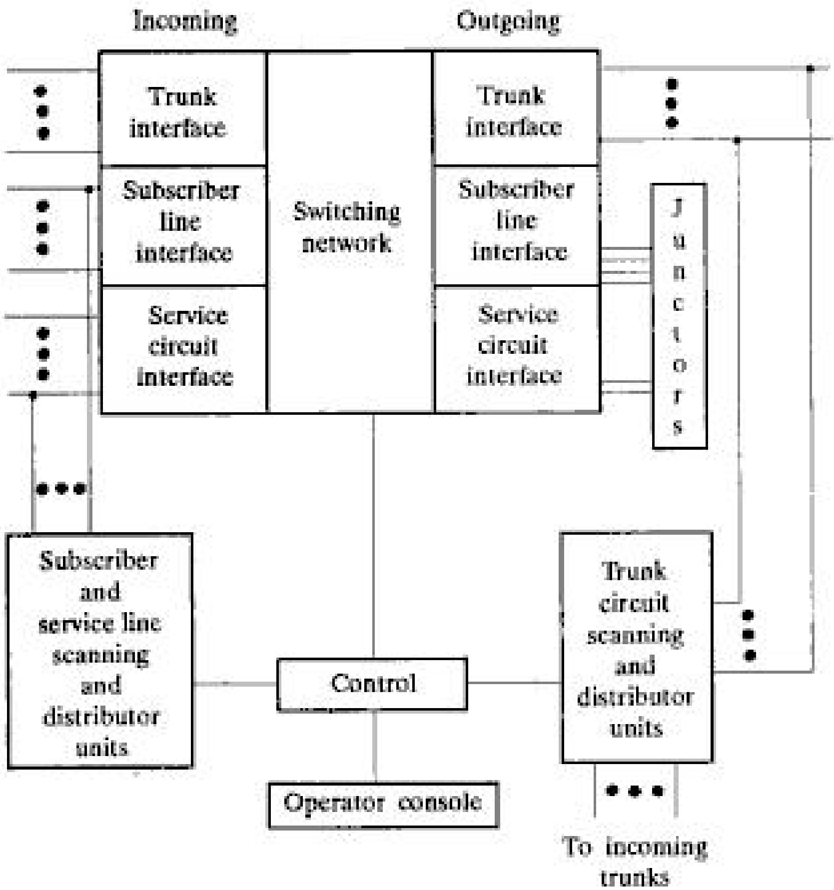


Fig. 1.10 Elements of a switching system.

provide an internal mechanism for local connections without using the junctor circuits. Line scanning units sense and obtain signalling information from the respective lines. Distributor units send out signalling information on the respective lines. Operator console permits interaction with the switching system for maintenance and administrative purposes. In some switching systems, the control subsystem may be an integral part of the switching network itself. Such systems are known as **direct control** switching systems. Those systems in which the control subsystem is outside the switching network are known as **common control** switching systems. Strowger exchanges are usually direct control systems, whereas crossbar and electronic exchanges are common control systems. All stored program control systems are common control systems. Common control is also known as indirect control or register control.

1.4 Manual Switching System

With the advent of automatic switching systems, the manual exchanges have almost gone out of use. Today, operator assistance is required on a routine basis, only to connect the incoming calls at a private automatic branch exchange (PABX) to the required extension numbers. Even this requirement will cease to exist with the large scale introduction of what is known as direct inward dialling (DID) facility which is described in Chapter 9. However, a discussion of the organisation of manual exchanges would help us to understand many of the principles of a telecommunication switching system.

As discussed in Section 1.2, a microphone requires to be energised in order to produce electrical signals corresponding to the speech waveform. In the very early switching systems the microphone was energised using a battery at the subscriber end. Later, a battery located at the exchange was used. Accordingly, one may place the early systems in two categories:

- Local battery (LB) exchanges
- Central battery (CB) exchanges.

In the LB systems, dry cells were used in subscriber sets to power the microphone. These cells have limited power output and cannot be used for signalling over long lines to the exchange. Hence, LB subscriber sets were provided with a magneto generator. In this case, a subscriber needed to rotate a handle to generate the required alternating current to operate indicators at the exchange. The use of magneto generator led to the alternative nomenclature **magneto exchange** for the LB systems. The necessity to replace dry cells frequently and the cumbersome procedure of rotating the magneto generator led to the development of CB exchanges, where a subscriber set is energised from a powerful central battery at the exchange.

Almost all the present day telephone exchanges are CB systems, although it is not inconceivable that future systems may resort to LB structures if low cost reliable power supplies for the subscriber premises become available. A simple CB system operated by a human being is shown in Fig. 1.11. The system consists of one or more switchboards manned by operators. The subscriber lines are terminated on jacks mounted on the switchboard. There is one jack for every subscriber line. Associated with each jack is a light indicator to draw the attention of the operator. When a subscriber lifts the hand set, the off-hook switch is closed, causing a current to flow through the hand set and the lamp relay coil. The lamp relay operates and the indicator corresponding to the subscriber lights up. The operator establishes contact with the subscriber by connecting the head set to the subscriber line via the headset key and a plug-ended cord pair. A plug-ended cord pair has two cords that are connected internally and terminated with a plug each at the external ends. The plugs mate with the jacks. To establish contact, a cord is plugged into the subscriber jack and the key corresponding to the chosen cord is thrown in position to connect the head set. On being

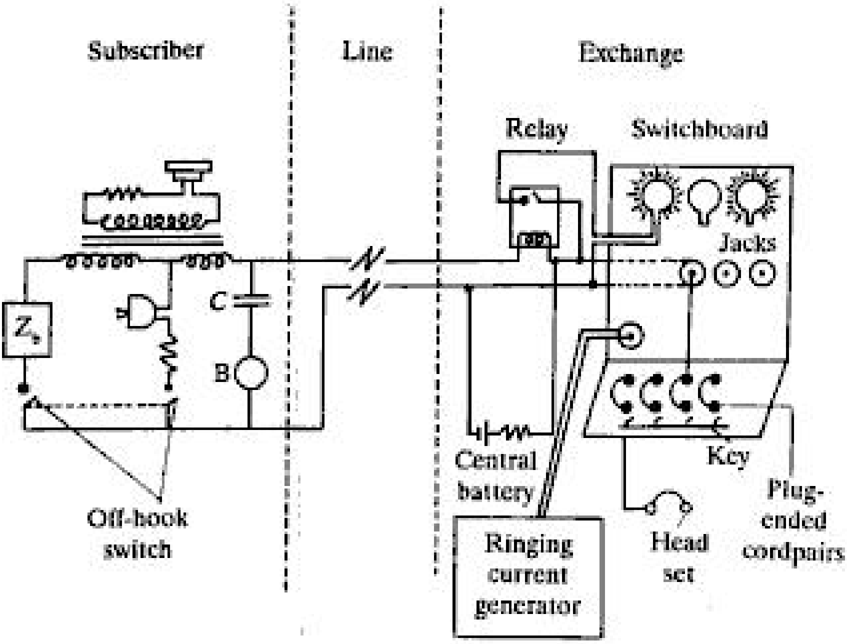


Fig. 1.11 Manned central battery exchange.

told the number required by the subscriber, the operator verifies whether the called party is free, and if so, sends out the ringing current to the called subscriber using a plug-ended cord pair. The ringing circuit at the subscriber end is usually a bell shown as B in Fig. 1.11, with a capacitor *C*, in series. They remain connected to the circuit always. The capacitor allows the alternating ringing current from the exchange to pass through the bell but prevents the loop direct current. If the called party is busy, the calling subscriber is told about the same. When the called party answers, his indicator lamp lights up. The operator then establishes a connection between the calling and the called party by plugging in the cord pair to the called party jack. In a manual switching system, the operator has full control of a connection. He enables the signalling systems, performs switching and releases a connection after a conversation.

If there are 200 subscribers terminated on a switchboard, there can be a maximum of 100 simultaneous calls. In order to support all these calls, the switchboard must contain 100 plug-ended cord pairs. But a single operator may not be able to handle 100 calls simultaneously. It is, however, rarely that all the subscribers would want to talk simultaneously. Assuming that only 20 subscribers (10 calls) will use the system simultaneously, the switchboard needs to be provided with only 10 plug-ended cord pairs. What happens if more than 20 users want to talk at the same time? The operator will not have

plug-ended cords for establishing the connection and the users are blocked. Users may also experience blocking, if the operator is not able to handle more than a certain number of calls simultaneously, even though free plug-ended cord pairs are available. In general terms, we may say that a user experiences blocking on account of the nonavailability of the switching circuits or the control system circuits.

When the number of subscribers increases, multiple switchboards and operators are required to handle the traffic. In this case, the subscriber switchboards at the exchange may be of two types:

- Single termination switchboards
- Multitermination switchboards.

The terms nonmultiple and multiple are sometimes used to denote single termination and multitermination schemes respectively. In the single termination scheme, a subscriber is terminated on only one board, whereas in the multitermination scheme, he is terminated on more than one switchboard. In single termination boards, subscribers are split into groups and connected to different switchboards. Each switchboard is handled by a separate operator. When a subscriber wishes to call another in the same group, the operator concerned establishes the call. In order to enable a subscriber belonging to one group to call a subscriber in another group, transfer lines are provided between the switchboards as shown in Fig. 1.12. The number of transfer lines is determined based on the estimated intergroup traffic. It may be noted that an intergroup call requires the services of two operators manning the two respective groups.

The maximum number of simultaneous calls within a group is limited by the number of plug-ended cord pairs in the group or the number of simulta-

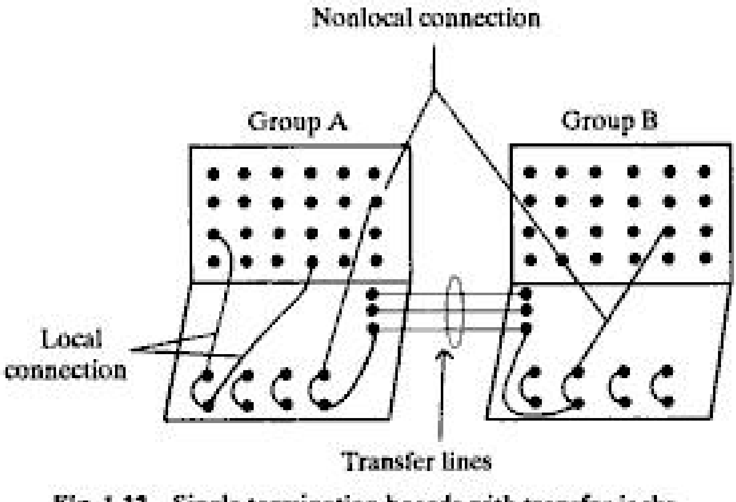


Fig. 1.12 Single termination boards with transfer jacks.

neous calls that can be handled by the operator, whichever is smaller. The number of simultaneous calls between the groups is limited by the number of transfer lines. Single termination systems suffer from the serious disadvantage that as many operators as there are switchboards are always required irrespective of the peak or lean traffic period. During lean traffic period, the average number of simultaneous calls is much less than that during the peak traffic period. Nevertheless, there are likely to be at least a few intergroup calls. Every intergroup call requires two operators to establish the call. Consequently, even a small number of intergroup calls among the switchboards demands that all boards be manned.

The need for two operators per call is avoided in the multitermination switchboard scheme. Here, every subscriber is terminated on all the switchboards as shown in Fig. 1.13. Such an arrangement has the advantage that a

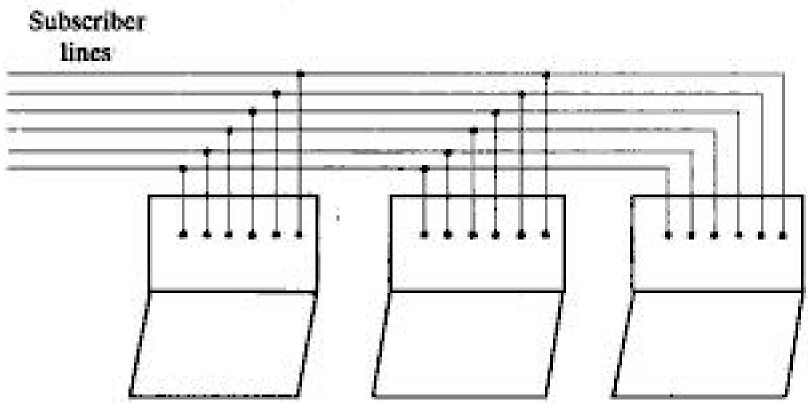


Fig. 1.13 Multitermination boards.

single operator can establish a call between any two subscribers connected to the system. The number of operators needed on duty at any time is now determined by the number of simultaneous calls estimated during the period. The system, however, has two drawbacks. Firstly, the total number of connections in the system increases considerably, thereby reducing the reliability of the system. Secondly, terminating all the subscribers in all the boards, such that the terminations are easily accessible to the operators, poses human engineering problems. The switchboard height becomes large, and the operator does not have easy access to the numbers at the top of the board. The problem is somewhat reduced by terminating half the number of subscribers in alternate switchboards in a cyclic manner and letting an operator have access to one-half of the adjacent boards on the left and right. The scheme is illustrated in Fig.1.14 for two operator positions. Subscribers are terminated on the boards as per the following order:

Subscriber Nos.	Operator board	Right/left Top/bottom
0–99	1	left-top
100–199	1	right-top
200–299	2	left-top
300–399	2	right-top
400–499	1	left-bottom
500–599	1	right-bottom
600–699	2	left-bottom
700–799	2	right-bottom

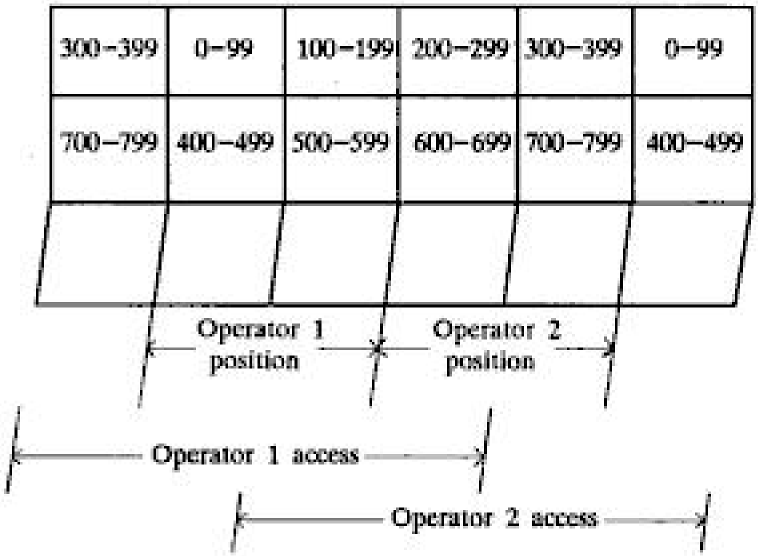


Fig. 1.14 A practical multitermination board scheme with cyclic assignment of numbers.

In addition, the numbers terminated on the right half of operator 2 panel, 300–399 and 700–799, are terminated on the half-panel on the left side of the operator 1. Similarly, the numbers terminated on the left half of operator 1 panel, 0–99 and 400–499, are terminated on the half-panel on the right side of the operator 2. Operator 1 gets access to numbers 300–399 and 700–799 from the left hand side half-panel and to numbers 200–299 and 600–699 from the half-panel of the operator 2. Similarly, operator 2 gets access to numbers 100–199 and 500–599 from the left and to numbers 0–99 and 400 – 499 from the right. It may be noted that we require one unmanned half-panel at either end of the switchboard row. Thus, in an 8-operator switchboard, there are nine logical switchboards.

As the number of subscribers increases, typically to a thousand or more, manual switching becomes more and more difficult and a method of automatic switching, signalling and control is inevitable.

1.5 Major Telecommunication Networks

Telecommunication networks may be categorised according to their coverage of geographical areas which have distinct telecommunication requirements. Towns and cities have high subscriber densities and relatively heavy traffic per subscriber. They are characterised by many local exchanges and short distances between exchanges as well as subscribers. Networks which are designed taking these factors into account are known as **urban** or **metropolitan** networks. Rural areas are characterised by low subscriber densities, widely dispersed subscribers, lighter traffic per subscriber, just one or two local exchanges usually far apart, long distances between subscribers and exchanges, and less conducive environmental conditions and inadequate infrastructural facilities. These areas are served by **rural networks**. **Long distance** or toll or **wide area networks** act as backbone networks interconnecting metropolitan and rural networks. They support intracountry, intercountry and intercontinental communications. Long distances (a few hundred to a few thousand kilometres) involved in such networks call for special consideration in the design of interexchange transmission and signalling systems. In the context of telephone networks, urban and rural networks are sometimes referred to as local networks. But in the context of data networks, the term local network refers to a network within a building or a campus.

The most stupendous telecommunication network in existence is the public switched telephone network (PSTN) or sometimes known as plain old telephone system (POTS). There are over 400 million telephones in the world and the length of wiring in the telephone network is estimated to be over 12 times the distance between the earth and the sun. The growth rate of the telecommunication industry is next only to that of the computer industry. But the growth value in absolute terms far exceeds that of the latter. The present trends seem to indicate that the growth rate may even surpass that of the computer industry in the 90s.

Telecommunication industry is both in the private and government or public sector. It is largely privatised in the United States where there are about 1600 privately owned telephone companies. In most of the other countries, the government has a complete monopoly over all forms of communications including mail, telegraph and telephone. Companies in the United States that provide communication services to the public are known as **common carriers**. In countries where the telecommunication authority is a nationalised company or a department of the government, it is usually known as the **post, telegraph and telephone** (PTT) administration. In India,

a single government department known as Posts and Telegraphs (P&T) department dealt with mail, telegraph and telephone communications until the end of 1984. With effect from January 1985, the responsibility was divided between two departments: the **Department of Posts** dealing with mail and the **Department of Telecommunications (DOT)** dealing with telephone, telegraph and data communications.

With a number of different agencies involved in providing telecommunication services, there is clearly a need to ensure compatibility on a worldwide scale to enable internetworking. This coordination is provided by the International Telecommunications Union (ITU), an agency of the United Nations. ITU has three main groups, one of which deals with telephone and data communications. This group is known by the French name: Comité Consultatif Internationale de Télégraphique et Téléphonique (CCITT), i.e. International Consultative Committee for Telegraph and Telephones. Five classes of members, A, B, C, D, and E constitute the CCITT. A description of the members belonging to different classes is presented in Table 1.1. Only class A members have voting rights. Since there is no PTT in the United States, the State Department represents the U.S. government as class A member. In CCITT terminology, a public telephone network is referred to as general switched telephone network (GSTN). In this text, we mostly use the popular term, PSTN; but when discussing standards, we use the CCITT term, GSTN.

Table 1.1 CCITT Members

Class	Members
A	National PTTs
B	Recognised private administrations
C	Scientific and industrial organisations
D	Other international organisations
E	Organisations whose primary function is in fields other than telecommunications, but have an interest in CCITT activities.

Data networks form the second major class of telecommunication networks. They are of recent origin (30–35 years old) and have emerged as a result of coming together of computer and communication technologies. These networks enable sharing of hardware, software and data resources of computer systems. As a result, they have come to be known as computer networks. Early networks interconnected computer systems of the same family. The network project, ARPANET, supported by the Advanced Research Projects Agency of the Department of Defence, United States, is one of the pioneering efforts in interconnecting heterogeneous systems. In some sense, the ARPANET demonstrated the feasibility of data/computer

networks. TYMNET is another large scale, general purpose data network introduced in 1970, interconnecting geographically distributed computer systems, users and peripherals.

Prompted by the developments in ARPANET and TYMNET, leading computer vendors announced proprietary architectures for interconnecting their own computer systems. These include IBM's System Network Architecture (SNA) and Digital Equipment Corporation's Digital Network Architecture (DNA). By 1980, the enormous value of computer communication networks became obvious, particularly among research communities and special interest user groups. The developers of Unix operating system quickly realised the advantages of networking and wrote a simple program called **uucp** (Unix-to-Unix Copy) for exchanging files and electronic mail among Unix machines. Two networks, UUNET (Unix Users' Network) and USENET have evolved based on **uucp** communication program. In addition to file transfer and electronic mail, USENET supports a service called **netnews**, which essentially provides a bulletin board on which any user may post a notice or news item to be seen by all other users of the network.

In the academic circle, the computer science community in the United States setup a network with the help of the National Science Foundation to serve all the computer science departments of the U.S. Universities. This network, known as CSNET, uses the transmission facilities of other networks but provides a uniform user level interface. Another academic network started by the City University of New York and Yale University, known as BITNET (Because It's Time Network) aims to serve all the departments of the universities. BITNET has now spread to a large number of sites and spans North America, Europe, Japan and Australia. In Europe, it is called European Academic Research Network (EARN). In the United Kingdom, there is a separate network known as Joint Academic Network (JANET) covering most of the universities and research laboratories.

As in the case of telephone networks, it is apparent that various agencies are involved in setting up and operating data networks and that there is a need for worldwide standards to enable data networks to interwork. Apart from CCITT, significant contributions to data network standards have come from the International Standards Organisation (ISO) which is a voluntary, nontreaty organisation. National standards organisations, like American National Standards Institute (ANSI), British Standards Institution (BSI), Association Francaise de Normalisation (AFNOR), Deutsches Institut für Normalische (DIN) and Bureau of Indian Standards (BIS) are members of ISO. The Institute of Electrical and Electronic Engineers (IEEE), the largest professional organisation in the world also plays a major role in evolving data network standards. Figure 1.15 presents the organisational structure of the different agencies involved in the coordination of telecommunication network activities.

Integrated services digital network (ISDN) is now emerging as a major telecommunication network. ISDN is envisaged as a single common net-

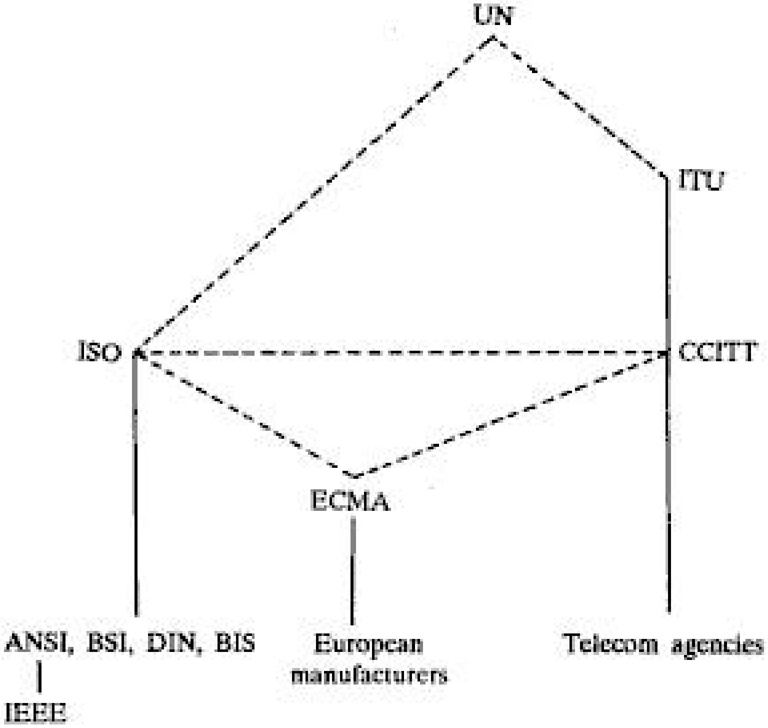


Fig. 1.15 Organisational structure of telecommunication coordination agencies .

work capable of carrying multimedia services like voice, data, video and facsimile. The key to ISDN is the digitalisation of services, transmission, switching and signalling. The digital domain acts as a common substratum for all current and future services. Once digitised, all signals, voice or nonvoice, look alike and a single digital network with adequate speed and signalling capabilities can support a wide range of services. However, meeting a variety of speed and signalling requirements is not easy. This is where the computers come in handy and the ISDN uses them extensively.

Recognising that the telephone network is the primary and extensive international communication infrastructure available today, ISDN is conceived to be a redesign of the existing telephone network to provide end-to-end digital connectivity. Obviously, the redesign cannot take place overnight and the ISDN will have to take an evolutionary path. At present, digital connectivity has been extended to user premises only in some parts of a few countries. Thus, ISDN will have to coexist with the present analog telephone network for some years to come. All these imply that the standards for ISDN must emerge well before its implementation. This aspect has been

recognised by CCITT and the first set of key ISDN recommendations were approved in 1984 and further refined in 1988. Unlike telephone and data networks, one may expect a fairly organised and structured growth in the case of ISDN, with CCITT spearheading the coordination.

Perhaps, ISDN is the single most important example of the contribution of computer technology to telecommunications and it may become the most important development as a result of the 'communion' of the computer and communication technologies. The large scale use of computers in ISDN is leading to the concept of **intelligent networks** which are preprogrammed to be adaptive, algorithmic, resourceful, responsive and intelligent. As an example of the possible capabilities of such intelligent networks, one may cite real time machine translation. A telephone conversation originating in Japanese may be heard in English at the receiver end and vice versa. A telex sent in Hindi in Delhi may be delivered in Kannada in Bangalore. Such examples, although somewhat far fetched now, may become a reality in the 21st century.

Telecommunication networks have been evolving in the last 150 years and would continue to evolve to provide wider services in a more convenient form in the coming century. The emerging information society depends heavily on the developments in the field of telecommunications. It is estimated that millions of dollars will be invested by many countries during the next two decades or so in improving the telecommunication facilities. We seem to be entering an era of 'sophisticated' telecommunications.

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EXERCISES

1. A fully connected network supports full duplex communication using unidirectional links. Show that the total number of links in such a network with n nodes, is given by $2 \times {}^{n}C_2$.

2. How are switching systems classified? In what way is stored program control superior to hard-wired control?

3. Estimate the bandwidth requirements of a single satellite that is to support 20 million telephone conversations simultaneously.

4. An electrical communication system uses a channel that has 20 dB loss. Estimate the received power, if the transmitted power is one watt.

5. If the signal input to an amplifier is 0 dBm, what is the power output in mW if the gain of the amplifier is 20 dB?

6. The channel interfaces in a point-to-point communication system attenuate the signal by 3 dB each. The channel has a loss of 30 dB. If the received signal is to be amplified such that the overall loss is limited to 20 dB, estimate the amplifier gain.

7. If the noise power in a channel is 0.1 dBm and the signal power is 10 mW, what is the (S/N) ratio?

8. What is the significance of (S/N) ratio being –3 dB?

9. For a carbon granule microphone, determine a suitable value for m, if the contribution from each of the higher order terms is to be less than $0.01\ I_0$.

10. What is the importance of a steady current flowing through a carbon microphone? Is the harmonic distortion affected by a change in tl energising current?

11. Why is it necessary to keep the magnetic diaphragm in an earphone displaced from its unstressed position? How is this achieved?

12. What happens if the ratio ϕ/ϕ_0 is not very small in the case of an earphone?

13. What is the significance of sidetone in a telephone conversation?

14. In the circuit of Fig.1.8, it is desired that 10 per cent of the microphone signal is heard as sidetone. If the number of turns in the coil P is 200, determine the number of turns in the coil Q and the secondary winding in the earphone circuit. Assume that Z_b is exactly matched to the line impedance on the exchange side.

15. In a 100-line folded network, how many switching elements are required for nonblocking operation?

16. A 1000–line exchange is partly folded and partly nonfolded. Forty per cent of the subscribers are active during peak hour. If the ratio of local to external traffic is 4:1, estimate the number of trunk lines required.

17. A central battery exchange is powered with a 48 V battery. The carbon microphone requires a minimum of 24 mA as energising current. The battery has a 400 Ω resistance in series for short circuit protection. The d.c. resistance of the microphone is 50 Ω. If the cable used for subscriber lines offers a resistance of 50 Ω/km, determine the maximum distance at which a subscriber station can be located.

18. A manual switchboard system needs to support 900 subscribers, numbered 100 – 999. Average peak hour traffic is 250 calls, 130 of which are within the number range 400 – 699, 20 of them are between this range and other range of numbers and the remaining are uniformly distributed in the other number ranges. The average lean traffic is 60 calls, of which no call is originated/destined from/to the number range 400 – 699 but uniformly distributed otherwise. An operator is capable of handling 30 simultaneous calls. Suggest a suitable manual switchboard system design that minimises the total number of terminations at the switchboards and employ the minimum number of operators. Estimate the number of terminations in your design.

2

Strowger Switching Systems

Strowger switching system was the first automatic switching system developed by Almon B. Strowger in 1889. The story goes that Strowger was an undertaker whose business seemed to have suffered on account of a telephone operator in a manual exchange. When subscribers rang up the operator and asked for an undertaker, she always connected them to her own husband who was also an undertaker and a competitor to A.B. Strowger. Annoyed at the amount of business he was losing this way, Strowger decided to make a switching system that would replace the human operator. The switch developed by him is named after him. Functionally, the system is classified as step-by-step switching system as the connections are established in a step-by-step manner.

Automatic switching systems have a number of advantages over the manual exchanges. A few important ones are:

- In a manual exchange, the subscriber needs to communicate with the operator and a common language becomes an important factor. In multilingual areas this aspect may pose problems. On the other hand, the operation of an automatic exchange is language independent.
- A greater degree of privacy is obtained in automatic exchanges as no operator is normally involved in setting up and monitoring a call.
- Establishment and release of calls are faster in automatic exchanges. It is not unusual in a manual exchange, for an operator to take quite a few minutes to notice the end of a conversation and release the circuits. This could be very annoying particularly to the business subscribers who may like to make a number of calls in quick succession.
- In an automatic exchange, the time required to establish and release a call remains more or less of the same order irrespective of the load on the system or the time of the day. In a manual system, this may not be true.

2.1 Rotary Dial Telephone

In manual exchanges, a calling subscriber may communicate the identity of the called subscriber in a natural and informal language to the operator. For example, a called subscriber may be identified by his name or profession or designation. In an automatic exchange, informal communication is not possible and a formal numbering plan or addressing scheme is required to identify the subscribers. Numbering plan, in which a subscriber is identified by a number, is more widely used than addressing scheme in which a subscriber is identified by alphanumeric strings. A mechanism to transmit the identity of the called subscriber to the exchange is now required at the telephone set. Two methods are prevalent for this purpose:

- Pulse dialling
- Multifrequency dialling.

Multifrequency dialling is discussed in Section 3.2. Pulse dialling originated in 1895 and is used extensively even today. In this form of dialling, a train of pulses is used to represent a digit in the subscriber number. The number of pulses in a train is equal to the digit value it represents except in the case of zero which is represented by 10 pulses. Successive digits in a number are represented by a series of pulse trains. Two successive trains are distinguished from one another by a pause in between them, known as the **interdigit gap**. The pulses are generated by alternately breaking and making the loop circuit between the subscriber and the exchange. The pulsing pattern is shown in Fig. 2.1 for digits three and two. The pulse rate is usually 10

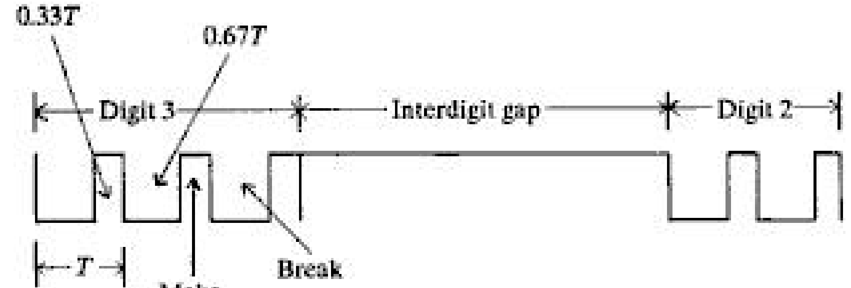


Fig. 2.1 Pulse dialling.

pulses per second with a 10 per cent tolerance. The interdigit gap is at least 200 ms although in some designs the minimum gap requirement may be as much as 400–500 ms. In some modern electronic and crossbar exchanges, there exists an upper limit for the interdigit gap (see Section 4.1). The duty ratio of the pulse is 33 per cent nominally.

In introducing dial pulsing mechanism in the telephone set, the following points have to be considered:

1. Since the pulses are produced by make and break of the subscriber loop, there is likelihood of sparking inside the telephone instrument.
2. The transmitter, receiver and the bell circuits of the telephone set may be damaged if the dialling pulses are passed through them.
3. The dialling habits of the users vary widely and hence all timing aspects should be independent of user action.

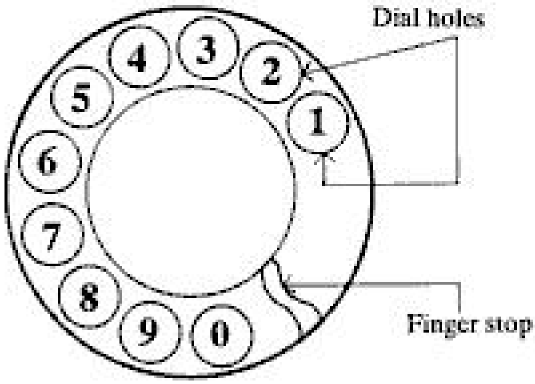
A rotary dial telephone uses the following for implementing pulse dialling:

- Finger plate and spring
- Shaft, gear and pinion wheel
- Pawl and ratchet mechanism
- Impulsing cam and suppressor cam or a trigger mechanism
- Impulsing contact
- Centrifugal governor and worm gear
- Transmitter, receiver and bell by-pass circuits.

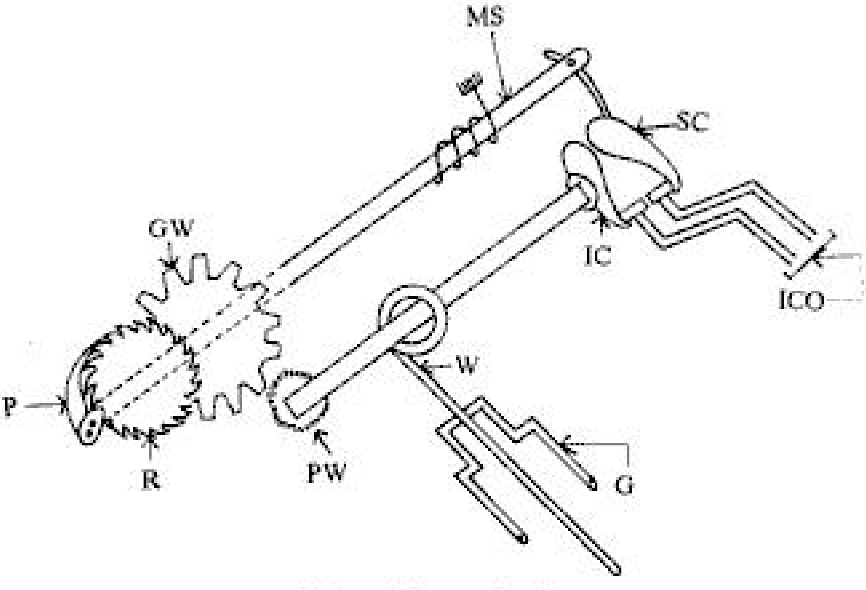
The arrangement of the finger plate is shown in Fig. 2.2(a). The dial is operated by placing a finger in the hole appropriate to the digit to be dialled, drawing the finger plate round in the clockwise direction to the finger stop position and letting the dial free by withdrawing the finger. The finger plate and the associated mechanism now return to the rest position under the influence of a spring. The dial pulses are produced during the return travel of the finger plate, thus eliminating the human element in pulse timings.

A rotary dial telephone is classified either as cam type or trigger type depending on whether a cam mechanism or a trigger mechanism is used for operating the impulsing contact. The general operating principle of both the types is the same and we explain the operation by considering the cam type. The internal mechanical arrangement of a rotary dial telephone is shown in Fig. 2.2(b). When the dial is in the rest position, the impulsing contacts are kept away from the impulsing cam by the suppressor cam. When the dial is displaced from its rest position, it is said to be in **off-normal** position. In this position, the impulsing contacts come near the impulsing cam. The rotation of the finger plate causes the rotation of the main shaft. The pawl slips over the ratchet during clockwise rotation. The ratchet, gear wheel, pinion wheel and the governor are all stationary during the clockwise movement of the dial. When the dial returns, the pawl engages and rotates the ratchet. The gear wheel, pinion wheel and the governor all rotate. The governor helps to maintain a uniform speed of rotation. The impulsing cam which is attached to a pinion shaft now breaks and makes the impulsing contacts which in turn causes the pulses in the circuit. The shape of the impulsing cam is such that the break and make periods are in the ratio of 2:1. When the dial is about to reach the rest position, the suppressor cam moves the impulsing contacts away from the impulsing cam. This action provides the required interdigit

gap timing independent of the pause that may occur between two successive digits, due to human dialling habit. Suppressor cam may also be designed



(a) Finger plate arrangement



(b) Impulsing mechanism

G = governo GW = gear wheel IC = impulsing cam
ICO = impulsing contact. MS = main shaft P = pawl
PW = pinion wheel R = rachet SC = suppressor cam
W = worm gear

Fig. 2.2 Rotary dial telephone parts and mechanism.

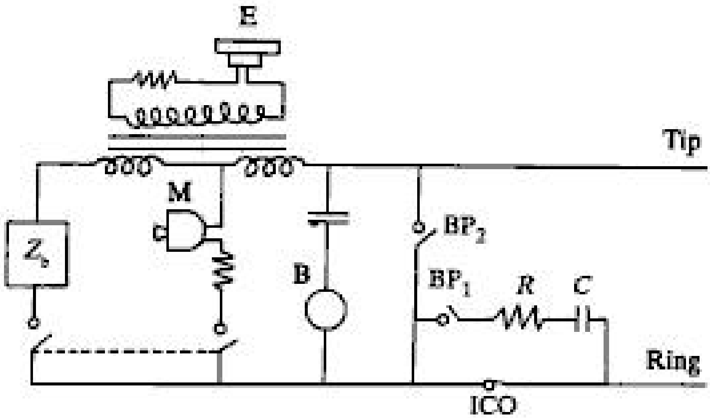
such that the interdigit pause is provided prior to the commencement of the first pulse of a digit.

The trigger dial is an improvement over the cam dial. The precision of operation in the cam dial is affected by the wear and tear of the cam elements and other friction members in the mechanism. The trigger dial design eliminates friction members and helps to achieve

- more uniform impulse ratio,
- larger interdigit pause, and
- better stabilisation of the return speed of the dial.

The trigger mechanism is so arranged that the trigger is sprung away from the impulse contacts during the clockwise motion of the dial, thus preventing pulsing at this stage. The trigger is sprung back to the operative position during the initial return motion of the dial and thereafter operates the pulse contacts. The time required to bring back the trigger to operative position provides the interdigit gap which is about 240 ms.

The impulsing circuit of the rotary dial telephone is shown in Fig. 2.3. When the subscriber lifts his handset (off-hook), the d.c. loop between the



B = bell BP = by-pass switch ICO = impulsing contact

Fig. 2.3 Impulsing circuit of a rotary dial telephone.

exchange and the subscriber is closed and a steady current flows through the loop. The impulsing contact (ICO), which is normally closed, is in series with the d.c. loop. When operated by the cam or the trigger, it breaks and makes the circuit. Figure 2.3 shows two by-pass switches BP_1 and BP_2. These switches close as soon as the dial is moved from its rest position and hence are known as dial-off-normal contacts. The switch BP_2 bypasses the microphone M, the earphone E, and the bell B, during pulsing. The switch BP_1 provides a local RC loop with ICO for quenching the spark that is produced

when the circuit is broken. In the absence of BP_1, the sparking voltage developed across ICO may affect adversely the other circuits in the telephone set. Once the dialling is complete, the dial is in the rest position, BP_1 and BP_2 are open, and the impulsing contact is closed. Thus the transmitter and the receiver are ready for speech conversation. The two wires connecting the telephone to the exchange are known as **ring and tip**. The central battery voltage of -48 V is connected through a relay to the ring lead and the tip lead is grounded. Ring lead is used to receive signals from the far end and the tip lead is used to transmit the signal.

2.2 Signalling Tones

As discussed in Section 1.4, a number of signalling functions are involved in establishing, maintaining and releasing a telephone conversation. These functions are performed by an operator in a manual exchange. In automatic switching systems, the verbal signalling of the operator is replaced by a series of distinctive tones. Five subscriber related signalling functions are performed by the operator:

1. Respond to the calling subscriber to obtain the identification of the called party.
2. Inform the calling subscriber that the call is being established.
3. Ring the bell of the called party.
4. Inform the calling subscriber, if the called party is busy.
5. Inform the calling subscriber, if the called party line is unobtainable for some reason.

Distinctive signalling tones are provided in all automatic switching systems for functions 1, 3, 4 and 5. A signalling tone for function 2 is usually not available in Strowger exchanges. However, most of the modern exchanges provide a call-in-progress or routing tone for function 2. Although attempts have been made to standardise the tones for various signals, many variations are in vogue in different parts of the world and even in different parts of the same country. Variations are mainly due to different capabilities and technologies of the switching systems used.

The signalling function 1 above is fulfilled by sending a **dial tone** to the calling subscriber. This tone indicates that the exchange is ready to accept dialled digits from the subscriber. The subscriber should start dialling only after hearing the dial tone. Otherwise, initial dial pulses may be missed by the exchange which may result in the call landing on a wrong number. Most often, the dial tone is sent out by the exchange even before the handset is brought near the ear. Sometimes, however, a few seconds may elapse before the dial tone is heard. This happens particularly in common control exchanges which use shared resources for user interfaces. The dial tone is a

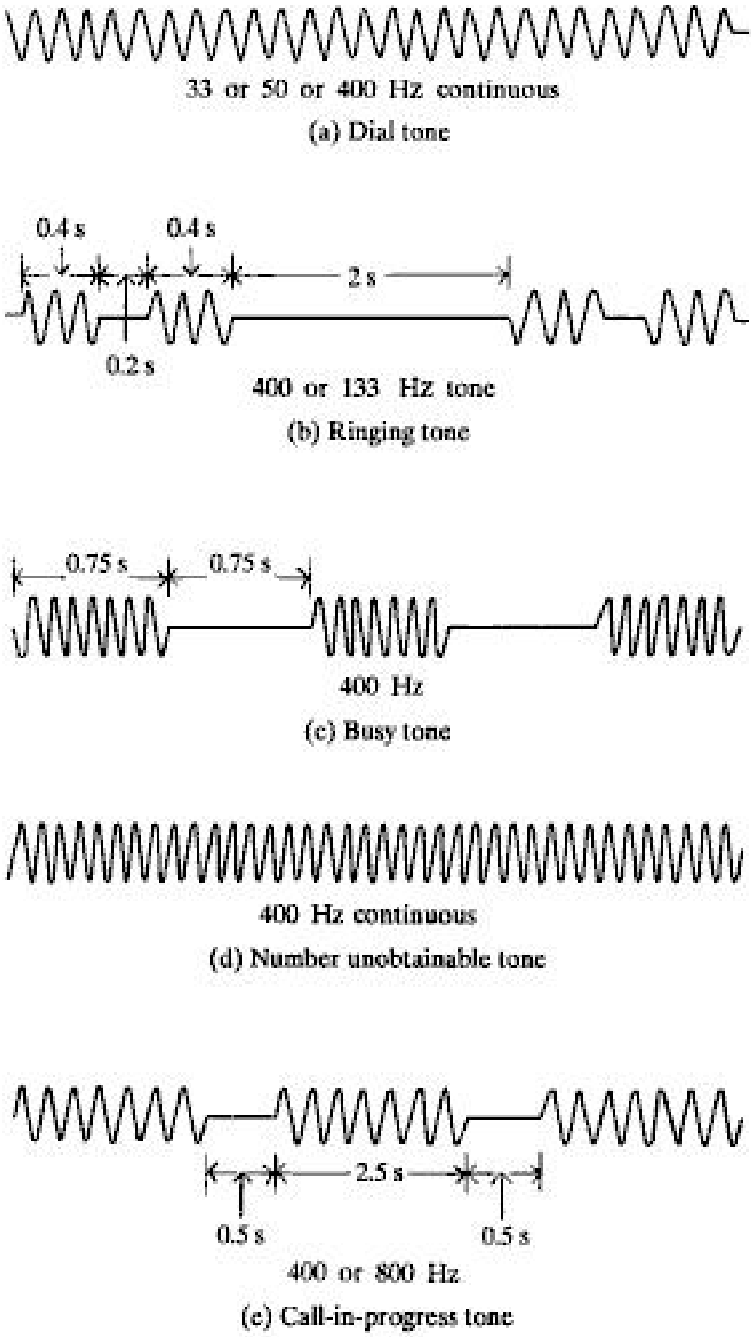


Fig. 2.4 Signalling tones in automatic exchanges.

33 Hz or 50 Hz or 400 Hz continuous tone as shown in Fig. 2.4(a). The 400 Hz signal is usually modulated with 25 Hz or 50 Hz.

When the called party line is obtained, the exchange control equipment sends out the ringing current to the telephone set of the called party. This ringing current has the familiar double-ring pattern. Simultaneously, the control equipment sends out a **ringing tone** to the calling subscriber, which has a pattern similar to that of the ringing current as shown in Fig. 2.4(b). The two rings in the double-ring pattern are separated by a time gap of 0.2 s and two double-ring patterns by a gap of 2 s. The ring burst has a duration of 0.4 s. The frequency of the ringing tone is 133 Hz or 400 Hz, sometimes modulated with 25 Hz or 33 Hz. It may be noted that the ringing current and the ringing tone are two independent quantities. This explains one of the common fault symptoms where a calling subscriber hears the ringing tone whereas no ring is heard at the called subscriber end.

Busy tone pattern is shown in Fig. 2.4(c). It is a bursty 400 Hz signal with silence period in between. The burst and silence durations have the same value of 0.75 s or 0.375 s. A busy tone is sent to the calling subscriber whenever the switching equipment or junction line is not available to put through the call or the called subscriber line is engaged. No distinction is made between these conditions. It is not possible for a calling subscriber to conclude on the basis of the busy tone that the called party was actually engaged in a conversation. While it is technically feasible to introduce different busy tones for different conditions, this would only, perhaps, confuse the subscriber, and not serve any useful purpose.

Figure 2.4(d) shows the **number unobtainable tone** which is a continuous 400 Hz signal. This tone may be sent to the calling subscriber due to a variety of reasons such as the called party line is out of order or disconnected, and an error in dialling leading to the selection of a spare line. In some exchanges the number unobtainable tone is 400 Hz intermittent with 2.5 s *on period* and 0.5 s *off period.*

The **routing tone** or **call-in-progress tone** is a 400 Hz or 800 Hz intermittent pattern. In electromechanical systems, it is usually 800 Hz with 50 per cent duty ratio and 0.5 s *on/off* period. In analog electronic exchanges it is a 400 Hz pattern with 0.5 s *on* period and 2.5 s *off* period. In digital exchanges, it has 0.1 s *on/off* periods at 400 Hz. When a subscriber call is routed through a number of different types of exchanges, one hears different call-in-progress tones as the call progresses through different exchanges. Figure 2.4(e) shows a routing tone pattern.

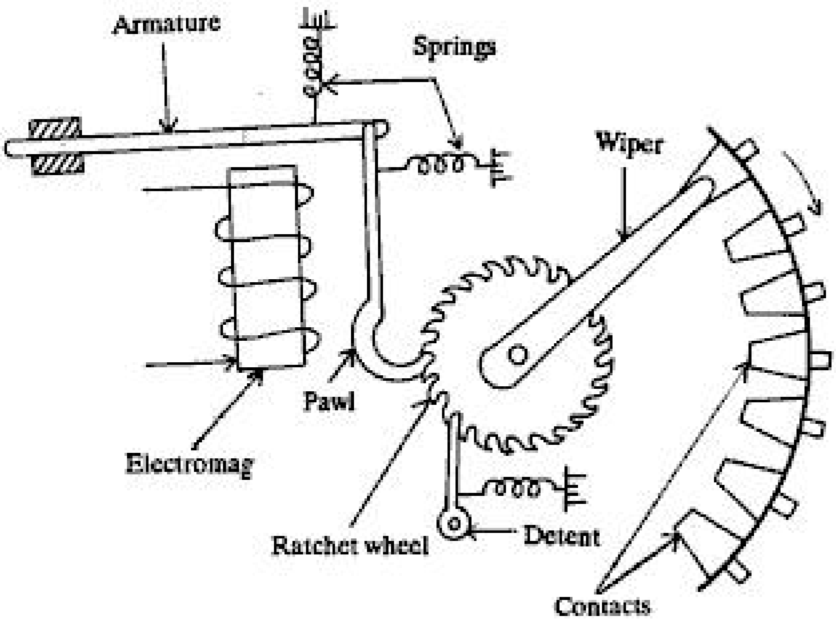
Regular users of telephone in a particular area have little difficulty in recognising signalling tones. It is not unusual that a subscriber in a new area where frequencies or timings of the tones are different from those in his own area, confuses signalling tones. In order to overcome this problem, recorded voices that announce messages like "number engaged" or "busy" are used in some modern exchanges. Voice announcement, however, poses problems in multilingual areas.

2.3 Strowger Switching Components

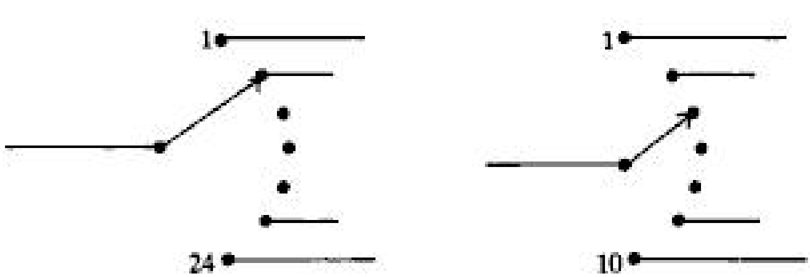
In the Strowger system, there are two types of selectors which form the building blocks for the switching system:

- Uniselector
- Two-motion selector.

These selectors are constructed using electromechanical rotary switches. The drive mechanism of a rotary switch is shown in Fig. 2.5(a).



(a) Drive mechanism of a rotary switch



(b) Schematic representation of uniselectors

Fig. 2.5 Uniselectors.

Whenever the electromagnet is energised, the armature is attracted to it and the pawl falls one position below the present tooth position. The ratchet wheel, however, does not move and is held in position by the detent. When the electromagnet is de-energised, the armature is released and returns to its rest position due to the restoring action of the spring. During this reverse motion of the armature, the pawl moves the ratchet wheel one position up where it is held in position by the detent. The clearance between the armature and the electromagnet is such that during the forward movement of the armature the pawl slips over the ratchet exactly by one position. As the ratchet wheel rotates up by one position, the wiper moves across one contact position in the direction indicated. Thus, if the electromagnet is energised and de-energised five times by applying five pulses, the wiper moves by five contacts. The mechanism shown in Fig. 2.5(a) is known as reverse drive type as the ratchet wheel moves when the armature returns to its rest position. It is possible to arrange the mechanism in such a way that the wheel moves during the forward motion of the armature in which case it is known as forward drive type. Reverse drive type is generally used in uniselectors and the forward drive type in two-motion selectors.

A uniselector is one which has a single rotary switch with a bank of contacts. Typically, there are four banks of which three are used for switching and the fourth one is used for homing. The three switching banks have 25 or 11 contacts each. The first contact in each bank is known as the home contact and the remaining as switching contacts. The homing bank has only two contacts: one at the first position corresponding to the home contacts of the other banks and the other extending as an arc from the second position to the last position. This arc contact is often referred to as the homing arc. Depending upon the number of switching contacts, uniselectors are identified as 10-outlet or 24-outlet uniselectors. Figure 2.5(b) shows a schematic representation of uniselectors.

The wipers associated with the banks of a uniselector, one for each bank, are rigidly mounted to a wiper assembly which moves whenever the ratchet wheel rotates. As a consequence, all the wipers move simultaneously and there is no relative motion amongst them. All wipers lie in the same vertical plane such that each wiper touches the same corresponding bank contact at any instant. There is an **interrupt contact** associated with the uniselector. This contact remains closed normally and opens whenever the armature is close to the end of its forward movement. It breaks the armature energising circuit to enable the armature to return to its rest position. It may be noted that if the drive circuit is permanently energised, the selector will step continuously owing to the constant breaking and making of the interrupter contact.

The proper functioning of a uniselector is dependent on a number of factors:

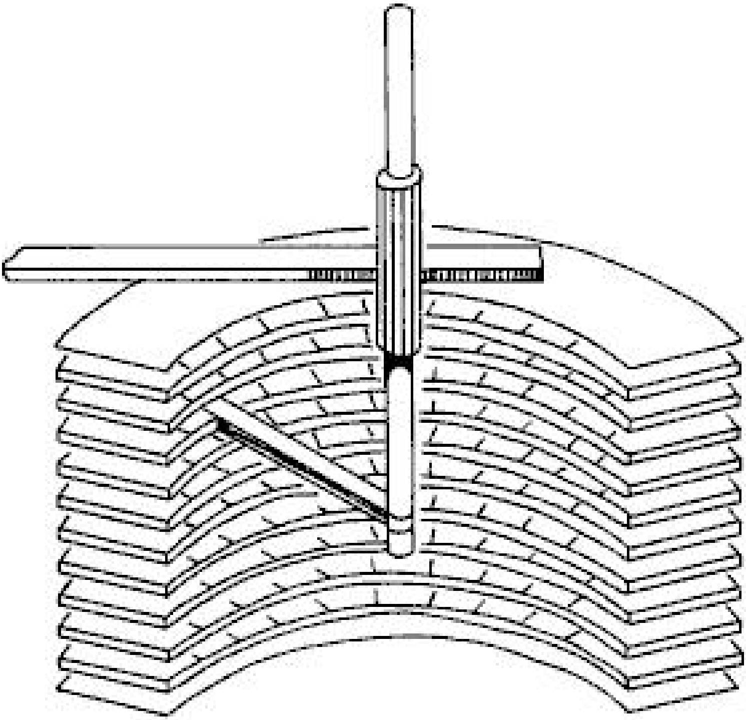
- Energising current level
- Inertia of the moving system

- Friction between wipers and bank contacts
- Friction in drive assembly
- Tension in restoring springs
- Adjustment of interrupter contacts.

To illustrate the importance of these factors for proper functioning, let us consider the adjustment of interrupter contacts as an example. The interrupter contacts must be adjusted so that they open and close at the correct instants in the stroke of the armature. If they open too soon, the armature may fail to complete its stroke and the pawl may not engage the next ratchet tooth. On the other hand, if they close too soon during the return of the armature, the reverse movement is affected and the stepping of the wiper assembly becomes uncertain. Wear and tear of the selector parts affect the proper functioning adversely and as a result, the selectors require frequent attention for maintenance.

A two-motion selector is capable of horizontal as well as vertical stepping movement. It has two rotary switches, one providing vertical motion for the wiper assembly, and the other providing horizontal movement for the wipers. Many authors use the term 'rotary switch' to mean the switch that causes horizontal movements of the wipers. In this text, the term rotary switch is used in a generic sense to imply a pawl and ratchet arrangement irrespective of whether such an arrangement is being used to cause vertical or horizontal motion. The horizontal movement rotary switch of a two-motion selector has an interrupter contact as in the case of uniselector. Normally, there are 11 vertical positions and 11 horizontal contacts in each vertical position. The lowest vertical position and the first horizontal contact in each vertical level are home positions, and the remaining ones are the actual switching positions. Thus, the wiper in a two-motion selector has access to 100 switching contacts. Access to any particular contact is obtained by moving the wiper assembly vertically to the required level and then rotating the wipers to the desired contact at that level. The arrangement is shown in Fig. 2.6 (a). At each level there are three or four banks of contacts. Depending upon the number of banks, a two-motion selector is sometimes known as a 330-point or 440-point selector. For homing the wiper assembly, it is driven beyond the 11th contact position by the horizontal rotary magnet and its interrupter contact. The wiper assembly then falls vertically to the home level and returns to the horizontal home position under the influence of a restoring spring. In some designs, a third magnet, known as **release magnet** is used for homing. A set of off-normal contacts are operated by the first vertical and horizontal movements of the wipers and they remain operated until the wiper assembly returns to home position. Figure 2.6(b) shows a schematic representation of a two-motion selector.

The vertical and horizontal motions in a two-motion selector may be effected directly by using two impulse trains from subscriber dialling. The first impulse train corresponding to the first digit operates the vertical mag-



(a) Two-motion selector arrangement

(b) Schematic representation

Fig. 2.6 Two-motion selectors.

net and the second impulse train drives the horizontal rotary switch. In such a case, it follows that the bank contacts are so numbered as to correspond to

the digits necessary to reach each contact. The numbering of a standard 100-contact bank is shown in Table 2.1. It may be noted that the lowest vertical level commences with 11 and ends with 10, whilst the tenth level commences with 01 and ends with 00. This is due to the fact that digit zero produces 10 pulses when dialled.

Table 2.1 Numbering Scheme for Two-Motion Selector Contacts

Level	Contacts									
	1	2	3	4	5	6	7	8	9	10
10	01	02	03	04	05	06	07	08	09	00
9	91	92	93	94	95	96	97	98	99	90
8	81	82	83	84	85	86	87	88	89	80
7	71	72	73	74	75	76	77	78	79	70
6	61	62	63	64	65	66	67	68	69	60
5	51	52	53	54	55	56	57	58	59	50
4	41	42	43	44	45	46	47	48	49	40
3	31	32	33	34	35	36	37	38	39	30
2	21	22	23	24	25	26	27	28	29	20
1	11	12	13	14	15	16	17	18	19	10

2.4 Step-by-Step Switching

A step-by-step switching system may be constructed using uniselectors or two-motion selectors or a combination of both. The wiper contacts of these selectors move in direct response to dial pulses or other signals like off-hook from the subscriber telephone. The wiper steps forward by one contact at a time and moves by as many contacts (takes as many steps) as the number of dial pulses received or as required to satisfy certain signalling conditions. Hence the name "step-by-step switching" is given to this method. Most of the necessary control circuits are built in as an integral part of the selectors, thus enabling them to receive and respond to user signalling directly. The relevant signalling tones are sent out to the subscriber by the switching elements (selectors) at the appropriate stages of switching. Thus, a step-by-step switching system is a **direct control** system.

A step-by-step switching system has three major parts as shown in Fig. 2.7. The line equipment part consists of selector hunters or line finders and the other two parts consist of selectors. The selector hunters and line finders represent two fundamental ways in which a subscriber gains access to common switching resources. As the name implies, a selector hunter searches and seizes a selector from the switching matrix part. There is one selector hunter for each subscriber. Usually, 24-outlet uniselectors are used as selector hunters. The selector hunter scheme is sometimes called subscriber uniselector scheme as there is a dedicated uniselector for each subscriber in the system. Line finders are associated with the first set of

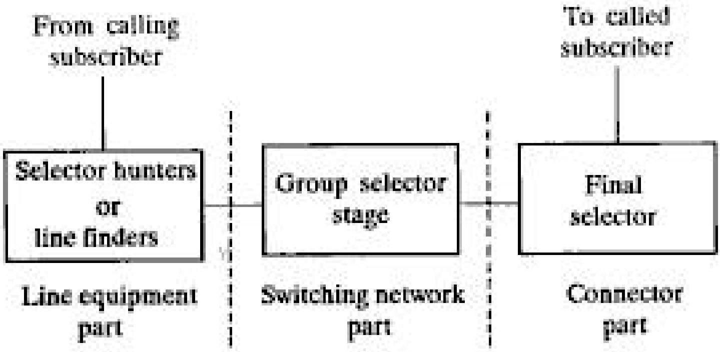


Fig. 2.7 Configuration of a step-by-step switching system.

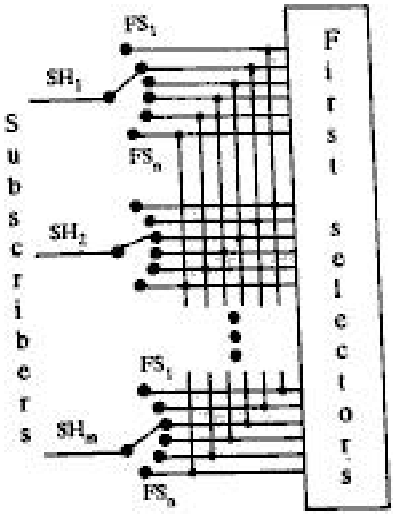
selectors in the switching matrix part and there is one line finder for each selector in the set. As the name implies, a line finder searches and finds the line of a subscriber to be connected to the first selector associated with it. Line finders are built using uniselectors or two-motion selectors. The line equipment part is also known as preselector stage. The selector hunters and line finders are generically referred to as preselectors.

The switching matrix part consists of one or more sets of two-motion selectors known as first group selector, second group selector, and so on. The larger the exchange size, the larger is the number of group selector stages. The connector part comprises one set of two-motion selectors known as final selectors. In small Strowger exchanges, all the parts may not exist. Configurations for different capacity exchanges are discussed in Sections 2.6–2.7.

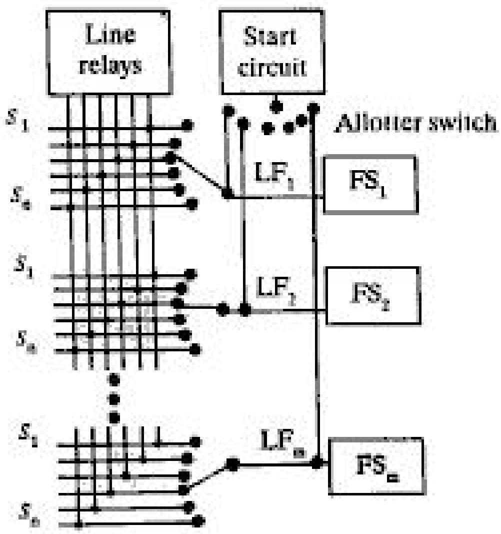
The selector hunter and line finder schemes are illustrated in the trunking diagrams shown in Fig. 2.8. In selector hunter based approach, when a subsciber lifts his hand set, the interrupter mechanism in his selector hunter gets activated and the wiper steps until a free first group selector is found at the outlet. The status of the first group selector, free or busy, is known by a signal in one of the bank contacts of the selector hunter. Once a free first selector is sensed, the interrupter is disabled and the first selector is marked 'busy'. Then, the first selector sends out a dial tone to the subscriber via the selector hunter which simply provides an electrical path. The first selector is now ready to receive the dialling pulses from the subscriber. It is possible that two selector hunters land on the same free first selector simultaneously and attempt to seize it. This is resolved by a suitable seizure circuit.

In the case of line finder based approach, the off-hook signal is sensed by all the line finders. Then the interrupter mechanism of one of the finders, whose associated first selector is free, gets activated and the line finder wiper steps until it reaches the contact on to which the subscriber is terminated. On finding the line, the concerned first selector sends out the dial tone to the subscriber in readiness to receive the dial pulses. The selection of one of the

line finders out of many free line finders, is achieved by means of an allotter switch in the start circuit of the line finder as shown in Fig. 2.8 (b). The circuit arrangements are such that the wiper of the allotter switch normally stands



(a) Selector hunter based access



(b) Line finder based access

FS = first selector LF = line finder SH = selector hunter

Fig. 2.8 Subscriber access to Strowger switching system.

on a contact connected to a free line finder and the first selector. When a subscriber lifts his receiver, the start signal from his relay is passed to the particular line finders via the common start circuit and the allotter switch. The line finder then commences to hunt for the calling line. As soon as the calling line is found, the allotter switch steps to the next free line finder. In effect, the line finder and the associated first selector to be used for the next future call is selected in advance by the allotter circuit. In practical designs, several allotter switches are provided in the system to serve calls that may originate in quick succession or simultaneously. Multiple allotter switches also avoid single-point failures, that might lead to complete breakdown of the system.

In designing large exchanges, some practical limitations are encountered in both the above schemes of gaining access to switching resources. Large exchanges are characterised by a large number of subscribers and first group selectors. It is not possible to provide a large number of outlets in the selector hunters or line finders such that any first group selector is accessible by any subscriber. Usually, subscribers are connected in groups of 100 to different sets of line finders which use two-motion selectors. Similarly, sets of selector hunters are connected to different groups of 24 first selectors each. Line finder and selector hunter approaches are advantageous for different sizes of the exchanges. If the exchange is small and the volume of traffic low, line finder approach is economical. For large exchanges with fairly heavy traffic, the selector hunter approach is more cost effective.

When the subscriber starts dialling, the first selector cuts off the dial tone and receives the pulse train corresponding to the first digit dialled by the subscriber. Its wiper assembly steps vertically as many steps as the number of dial pulses. The wipers then move in the horizontal plane across the contacts until they come across a contact to which a free second group selector is connected. This horizontal stepping is completed within the interdigit gap of about 240 ms. Thereafter, the first group selector just provides an electrical path to the second group selector. Each group selector stage functions in a fashion similar to the first group selector by processing one digit of the number dialled by the subscriber and finding a group selector in the subsequent stage. The last two digits of the dialled number are processed by the final selector which steps vertically according to the last but one digit and steps horizontally according to the last digit. Since the final selector responds to digits both in the vertical and horizontal directions unlike the group selectors, it is sometimes referred to as **numerical selector**. If the called subscriber is free, as sensed from a signal at the corresponding bank contact, the final selector sends out a ringing current to the called subscriber and a ringing tone to the calling subscriber. When the called subscriber lifts his handset, the ringing current and tone are cut off and the call metering circuits are enabled by the control circuits associated with the final selectors. If the called subscriber is busy, the final selector sends out a busy tone to the calling subscriber. At any stage of switching, if there is no

free selector at the next stage, a busy tone is returned to the calling subscriber.

The control functions in a Strowger system are performed by circuits associated with the selectors. Control and supervisory signals are carried from stage to stage by means of contacts in one of the banks. The wire interconnecting these banks is known as *P*-wire or private wire. Two other bank contacts are used for carrying voice signals and the associated wires are known as negative and positive wires which extend up to the subscriber premises. A selector *X* is said to have seized another selector *Y* in the next stage when the negative, positive and private wires of the selector *X* have been connected to the negative, positive and private wires respectively of the selector *Y*. The complexity and functionality of the control circuits associated with a selector vary depending on the position of the selector in the switching stage.

All the selector control circuits are composed of one or more of the following basic circuits:

1. Guarding circuit
2. Impulsing circuit
3. Homing circuit
4. Metering circuit
5. Ring-trip circuit
6. Alarm circuit.

The guarding circuit is an essential feature of all the selectors. It guards the selector by making it busy as soon as it is seized, lest some other selector involved in setting up another call may also seize it. Once applied, the guarding condition remains set as long as the call is not terminated. The guarding condition is indicated by an earth on the *P*-wire. An earth is supplied to the *P*-wire by the home contact and the homing arc of the home bank. To avoid any unguarded period during the transition of the wiper from the home contact to the homing arc, the wiper is of **bridging** type, i.e. it functions in **make-before-break** fashion; it touches the homing arc before it leaves the home contact.

The impulsing circuit is an essential part of all those selectors which have to respond to dialling pulses. It is used in group and final selectors, but not in line finders or selector hunters. This circuit is usually designed around three relays: one fast acting and the other two slow acting. The fast acting relay faithfully responds to the impulses and passes them on to the *P*-wire circuit. The fast action is achieved by using only one contact spring assembly and an isthmus armature. One of the slow acting relays serves to maintain guarding conditions on the *P*-wire of the incoming circuits and provides for the connection of the selector magnet to the impulsing relay. The third relay is used to recognise the end of a pulse train corresponding to a single digit and prepare the circuit for the next stage in the switching process.

When a selector is searching for a free outlet, the condition on the *P*-wire must be tested to determine whether the outlet is free or not. If an outlet is engaged, the wipers must be allowed to continue the hunting process. If the outlet is free, it must be seized immediately and the incoming positive and negative wires must be switched through to the input of the next stage. At the same time, the hunting process must stop. Once established, the connections must be held until the conversation lasts. All these functions are performed by the testing circuit, and hence this circuit is sometimes referred to as hunting, testing, switching and holding circuit.

There are two methods of indicating the free condition on the *P*-wire: one by means of a simple disconnection and the other by applying a battery to the *P*-wire. As mentioned earlier, the busy condition is indicated by an earth connection. Hence, a testing circuit has to distinguish between an earth and a disconnection in one case and between an earth and a battery in the other. Accordingly, the two methods are referred to as **earth testing** and **battery testing**, respectively. Battery testing is less prone to false connections than earth testing. In any switching process, particularly electromechanical switching, momentary disconnections of lines do occur. Therefore, false switching may take place if the earth testing happens at an instant when a busy outlet is in the course of some switching or release process which temporarily disconnects the guarding earth from the *P*-wire. Such a problem does not occur in the case of battery testing.

At the end of a conversation, all the selectors used for the call must be released and returned to their respective home positions. This operation is performed by homing circuits. The two-motion selectors return to their home position by actuating their self-drive mechanism using interrupt contact. In the case of uniselectors, the necessity of homing arises only for the calling subscriber uniselector. The called subscriber uniselector is already in the home position. Homing operation requires a finite time, and it must be ensured that a hunting selector may not seize a selector which is in the process of homing. Thus, the provision of guarding earth during homing is an integral feature of the homing circuit.

Metering circuit is a special feature of the final selectors. It registers a call against the calling party as soon as the called party answers. The circuit drives a meter containing a simple ratchet-operated counting mechanism with a capacity of 4 to 5 digits. For local calls, the metering is usually independent of the duration of the call and the meter is pulsed only once by the final selector. For long distance calls established using subscriber trunk dialling (STD) facility, the metering is time dependent and the meter is pulsed at an appropriate rate. In this case, the metering pulses are usually received from a remote exchange. Metering is achieved by connecting the meter to the *P*-wire of the subscriber uniselector through a rectifier and applying a positive voltage which makes the rectifier conduct and thereby pulse the meter. The use of the rectifier also ensures that *P*-wire remains guarded during metering.

Ring-trip circuit is a part of the final selectors. The attention of the called subscriber is drawn by ringing the bell of his telephone set. At the same time, a ringing tone is sent out from the final selector to the calling subscriber. Both the ringing current and the ringing tone are cut off by the ring-trip circuit as soon as the called party answers the call. The ringing current in a Strowger system is a 17 Hz alternating current. The ringing tone and the period of interruption of the ringing current are controlled by a relay which is driven by suitable pulsing circuits. To prevent the ringing current from interfering with the speech circuit, the electrical power to the ringing circuit is isolated from the main exchange supply. As soon as the condition of main power being applied to the circuit is sensed, the ringing current is tripped. A common fault of premature tripping of the ringing current occurs when the main supply battery gets connected to the circuit during ringing without the called subscriber actually lifting the handset. If this happens, the bell at the called subscriber telephone set rings only once or twice.

Alarm circuits provide visual and audible indications of any fault or undesirable condition creeping into the selector circuits. Three types of faults are usually detected: **off-hook condition, called-subscriber-held,** and **release held**. In the event of a short-circuit in the subscriber line or the subscriber not having replaced his handset properly on the hook, his d.c. loop circuit remains closed and his uniselector hunts and seizes a first selector unnecessarily. To avoid this undesirable use of power and switching elements, every first selector is provided with a permanent glow alarm circuit. This circuit activates an audio and a visual alarm if a selector remains seized for more than six minutes. Called-subscriber-held alarm circuit is necessary in all exchanges where the release of the switching stages is initiated by the calling subscriber replacing his handset. In case the handset is not properly replaced, all the selectors and the called subscriber line remain held, even though the called subscriber has replaced his handset properly. If this happens, neither the called subscriber is able to make any call himself nor can anybody else call him. Thus, the subscriber's instrument remains paralysed. A miscreant can easily create this situation by calling a number and then not replacing his handset on the hook. To prevent this, all final selectors are provided with called-subscriber-held alarm circuit. If the condition of the called subscriber handset having been replaced and the calling subscriber handset not having been replaced lasts for over three minutes, this alarm circuit operates. The third type of alarm circuit, i.e. release-held alarm circuit, senses the failure of a selector to return to home position.

2.5 Design Parameters

When considering the design of a switching system, a number of design alternatives and options may be available. For example, a Strowger switching system may be designed entirely on the basis of uniselectors or two-motion selectors, or a combination of both. It then becomes necessary to compare

and evaluate designs to choose from the alternatives. Design parameters assist us in this process. In this section, we define a set of design parameters that characterise the switching systems. These parameters are generic in nature and hence are applicable to all types of switching systems irrespective of the technology or architecture.

The switching network is a major component of any switching system. It is mainly composed of switching elements and the associated circuits. As a result, the cost of the switching network is directly proportional to the number of switching elements in the network. Hence, a good design must attempt to minimise the number of switching elements in the system. When considering the total switching systems, there are other cost elements. For common control systems, the cost of the control subsystem must be taken into account. There is a cost associated with some fixed common hardware elements like ringing current generator, different tone generators and power supplies. A switching network may be realised using one or more stages of switching elements. The higher the number of stages, the longer is the time taken to set up a call as switching is involved in every stage. Every switching system is designed to support a certain maximum number of simultaneous calls, which we call as the **switching capacity**. In most of the designs, the entire switching resources are not utilised even when the switching capacity is fully utilized. Part of the resources remains idle. The fraction of the hardware actually used under full load conditions is an index of the design. Taking these factors into account, we now enumerate the design parameters:

1. Number of subscriber lines, N
2. Total number of switching elements, S
3. Cost of the switching system, C

$$C = S \times C_s + C_c + C_{ch}$$

where

C_s = cost per switching element

C_c = cost of the common control system

C_{ch} = cost of the common hardware

Since the control circuits are associated with switching elements in a Strowger system, C_c is equal to zero. The common hardware is usually a small proportion of the total hardware except for the power supplies and its cost is of the same order in different comparable designs. Hence, we ignore C_{ch} in most of our calculations.

4. Switching capacity, SC
5. Traffic handling capability, TC

$$TC = \frac{\text{switching capacity}}{\text{theoretical maximum load}}$$

$$= \frac{2(SC)}{N}$$

6. Equipment utilisation factor, *EUF*

$$EUF = \frac{\text{number of switching elements in operation when the } SC \text{ is fully utilised}}{\text{total number of switching elements in the system}}$$

7. Number of switching stages, K
8. Average switching time per stage, T_{st}
9. Call setup time, T_s

$$T_s = T_{st} \times K + T_0$$

where T_0 is the time required for functions other than switching. T_0 is a significant quantity in common control systems where control functions are separated from switching functions. In Strowger (direct control) systems, T_0 may be ignored.

10. Cost capacity index, *CCI*

$$CCI = \frac{\text{switching capacity}}{\text{cost per subscriber line}} = \frac{N(SC)}{C}$$

The higher the value of *CCI*, the better is the design. Given the traffic handling capability of a switching system, the stochastic behaviour of the actual traffic and holding time characteristics of a call, it is possible to make reasonable estimates of the blocking probabilities. A detailed treatment of the blocking behaviour of switching systems is presented in Chapter 8. However, simple blocking probability calculations are made when the designs are discussed in the earlier chapters. It may be noted that the blocking probability is more of a performance parameter than a design parameter. However, at the design stage, the traffic handling capability of the switching system must be sized to achieve a low blocking probability in the field. This is done on the basis of estimated traffic.

2.6 100-line Switching System

A 100-line switching system can serve up to 100 subscribers. A 100-line Strowger switching system may be configured in a variety of ways. In this section we discuss five different design alternatives for a 100-line step-by-step switching system. We then compare the designs based on the design parameters discussed in Section 2.5. Simple line diagrams known as **trunking diagrams** are used to represent the configurations of switching systems. For computing the cost of different designs, we assume that the cost of a uniselector is one unit and that of the two-motion selector is two units.

2.6.1 Design 1

An elementary configuration for a 100-line Strowger switching system using 10-outlet uniselectors is shown in Fig. 2.9. The configuration has two stages.

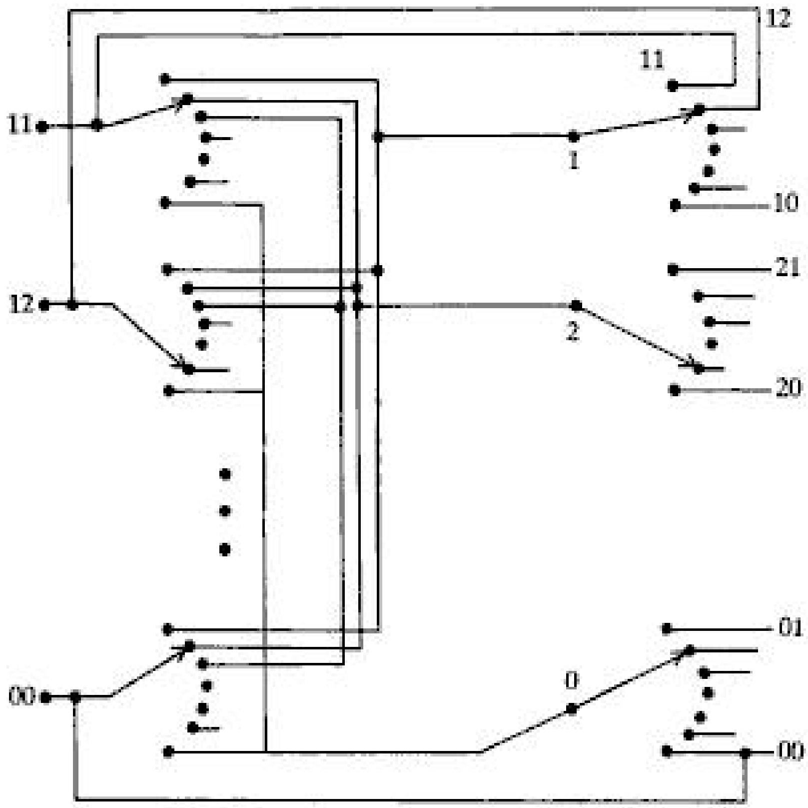


Fig. 2.9 100-line switch using uniselectors

In the first stage, there are 100 uniselectors, one for each subscriber. The second stage has 10 or more uniselectors. The second stage outlets are folded back to the corresponding inlets via suitable control circuitry (not shown in the figure for the sake of simplicity). Usually, each subscriber line is terminated on a relay group at the exchange. The relay group contains all the necessary circuits for the control of the switching mechanism. Functions like testing, switching and return of the tones are done by the relay groups. Similarly, outlets from the first stage are terminated on relay groups at the input of the second stage. The four banks of the uniselectors serve to provide positive, negative, *P*-wire and homing connections. The corresponding outlets of all the first stage uniselectors are commoned or multipled. The first stage responds to the first digit dialled by the user and the second stage to the

second digit. Suppose the subscriber 12 dials the number 56, his uniselector, i.e. uniselector 12, steps by five positions and the uniselector 5 in the second stage steps by six positions. With 10 uniselectors in the second stage, only 10 calls can be established simultaneously. Even this requires that the calls are uniformly distributed one per decade throughout the number range. All calls in a given decade use the same second stage uniselector. For example, the numbers 50–59 are put through uniselector number 5 in the second stage. As a result, two calls destined for numbers in the range 50–59 cannot be put through simultaneously, even though other uniselectors may be free in the second stage. This problem may be overcome by making such an arrangement by which the uniselectors in the second stage are treated as a common resource for all the uniselectors in the first stage. The design parameters for this design are:

$$S = 110, \quad SC = 10, \quad K = 2, \quad TC = 0.2,$$
$$EUF = 0.18, \quad C = 110, \quad CCI = 9.09.$$

In this design, blocking may occur under two conditions:

1. The calls are uniformly distributed, 10 calls are in progress and the 11th one arrives.
2. The calls are not uniformly distributed, a call is in progress and another call arrives, which is destined for a number in the same decade.

The blocking probability P_B in the first case is dependent upon the traffic statistics. If we assume a random distribution of calls in the second case, we can calculate P_B as

Probability that there is a call in a given decade = 10/100

Probability that another call is destined to the same decade but not to the same number = 9/98

Therefore,

$$P_B = (1/10)(9/98) = 0.009$$

2.6.2 Design 2

An alternative scheme which does not involve any logic circuit is to employ 10 uniselectors in the second stage for every one uniselector in the first stage. The total number of uniselectors in the system becomes 1100; 100 in the first stage and 1000 in the second stage. There are 10,000 outlets and 100 inlets. The corresponding outlets associated with every inlet are commoned. For example, all outlets numbered 10 are commoned together. Thus, effectively there are only 100 independent outlets from the switch which are folded back to the corresponding inlets. It may be noted that unlike the previous design, this switching system is nonblocking. The design parameters are:

$$S = 1100, \quad SC = 50, \quad K = 2, \quad TC = 1,$$
$$EUF = 0.09, \quad C = 1100, \quad CCI = 4.54, \quad P_B = 0.$$

Some observations are in order. Apparently, Design 1 appears to have serious limitations. But the values of design parameters, *CCI*, *EUF* and P_B indicate that it is more cost effective than Design 2. If the traffic statistics indicate that more than 10 calls originate most of the time, the blocking performance of Design 1 becomes unacceptable. In cases where the average number of calls exceeds 10 but still a small fraction (say, less than 20) of the theoretical maximum number of calls, a via media configuration with more than one uniselector per decade in the second stage would be a good solution. But this also calls for a mechanism to choose a free selector out of the many available at the second stage. In step-by-step switching systems, the selection of one out of many selectors in the next subsequent stage is done by deploying a uniselector or the horizontal rotary mechanism of a two motion selector in a self-stepping mode using the interrupter contacts. Designs 4 and 5 discussed later in this section use such arrangements.

2.6.3 Design 3

Another way of realising a 100-line Strowger switching system is to use one two-motion selector for each subscriber. A subscriber is assigned a number in the range 00–99, and the same number is used to identify the two-motion selector assigned to him. The 100 outlets of each two-motion selector are numbered as per the scheme given in Table 2.1. The corresponding outlets in all the 100 two-motion selectors are commoned and folded back to the corresponding inlets. For example, a subscriber with 67 as his number is assigned the two-motion selector 67. The outlet 67 which corresponds to this subscriber is connected to the 7th contact in the 6th vertical position of all the two-motion selectors and folded back to his inlet. The arrangement is shown in Fig. 2.10. If subscriber 23 dials 67, his two motion selector 23 would step vertically 6 times corresponding to the first digit and would step horizontally 7 times to reach the contact to which the subscriber 67 is connected. This switch is nonblocking and uses only one stage of switching elements.

The two-motion selector used to establish a call is dependent upon the initiator of the call. For example, when 23 calls 45, the two-motion selector 23 is used, whereas when 45 calls 23, the two-motion selector 45 is used, although the parties in conversation are the same in both the cases. Since the two-motion selector is activated by the calling party, the call is terminated only when the calling party disconnects the line. If a two-motion selector goes out of order, the subscriber connected to it will not be able to make any outgoing calls but can receive incoming calls. The design parameters of this switch are:

$$S = 100, \quad SC = 50, \quad K = 1, \quad TC = 1,$$
$$EUF = 0.5, \quad C = 200, \quad CCI = 25, \quad P_B = 0.$$

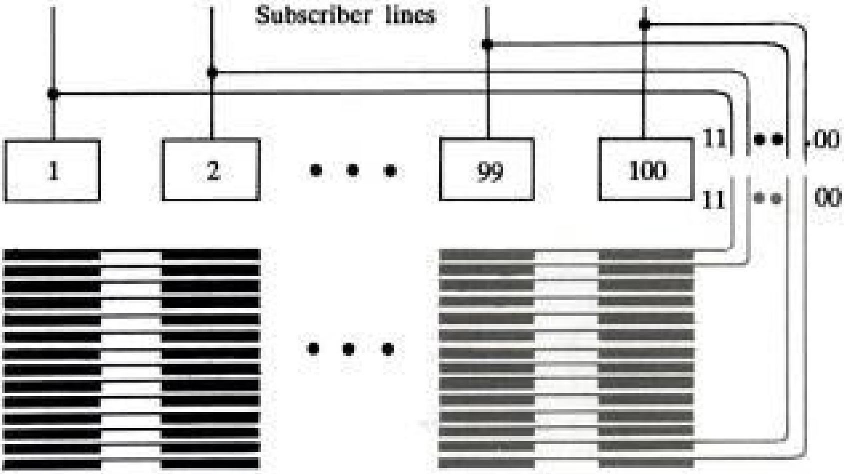


Fig. 2.10 100-line exchange with one two-motion selector per subscriber.

Clearly, Design 3 is superior to Designs 1 and 2. Further improvements to Design 3 are possible if the switching capability is provided to meet only the estimated peak-hour load rather than the theoretical maximum load. Such a design would demand that the switching elements be treated as a common resource accessible by all the subscribers. An elementary requirement of such a design is that the cost savings resulting from placing the switching elements in a common pool should be greater than the cost of the equipment required to associate a subscriber line with a selector. Secondly, the time taken to associate a selector to the subscriber line should not be excessive, and the dial tone must be returned to the subscriber without appreciable delay. Finally, the design must not unduly complicate the maintenance of the equipment and must provide a ready means for tracing connections. Designs 4 and 5 treat switching elements as a common resource.

2.6.4 Design 4

Instead of 100 two-motion selectors as in the case of Design 3, let us consider only 24 two-motion selectors. In this case, 24 simultaneous calls can be put through the switch. The 24 two-motion selectors are shared by all the hundred users. The corresponding outlets of the two-motion selectors are commoned as in the previous case. It is implicitly assumed here that the average peak-hour traffic is 24 simultaneous calls.

We now have to introduce a mechanism by which a user can get hold of a two-motion selector whenever he wants to make a call. Once he seizes a two-motion selector, obtaining the required number follows the same procedure as in the case of Design 3. As discussed in Section 2.4, we may adopt

either selector hunter or line finder approach. In this design, we use selector hunters and in Design 5 (see section 2.6.5), we use line finders.

Typically, a 24-outlet uniselector is used as a selector hunter. Each of the 24 outlets is connected to one two-motion selector. Thus, a subscriber has access to all the 24 two-motion selectors in the system. The corresponding outlets of all the selector hunters are commoned and thus, all subscribers have access to all the two-motion selectors. This scheme is shown in Fig. 2.11.

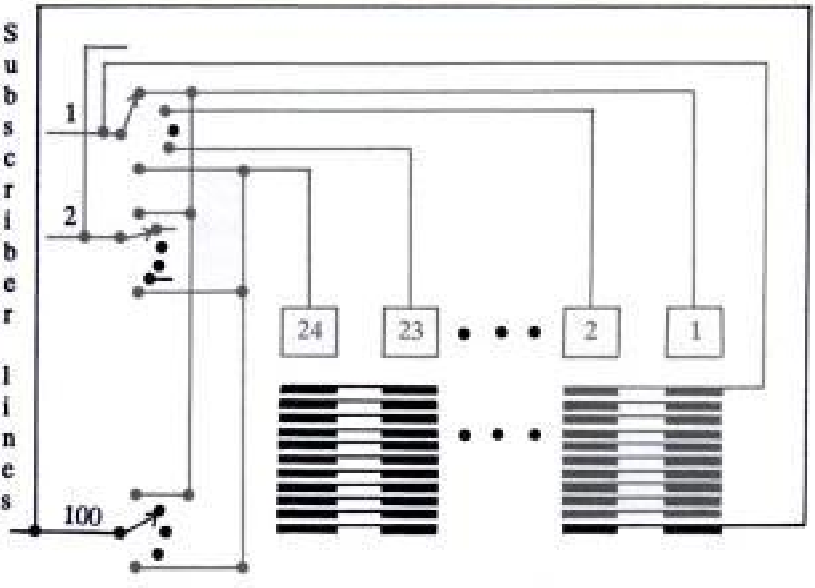


Fig. 2.11 100-line exchange with selector finders.

The call establishment in this scheme takes place in two steps. Firstly, when the subscriber lifts his receiver handset, his uniselector hunts through the contact positions and seizes a free two-motion selector. At the next step, the two-motion selector responds to the dial pulses for appropriate connection. The design parameters of this system are:

$$S = 100 \text{ uniselectors} + 24 \text{ two-motion selectors}$$

$$SC = 24, \quad K = 2, \quad TC = 0.48,$$
$$EUF = 0.58, \quad C = 148, \quad CCI = 16.2.$$

The blocking probability would depend on the traffic characteristics. For an exchange with 100 subscribers, the probability of more than 48 subscribers being active simultaneously is very low. Hence, blocking performance of this design must be satisfactory. This design is clearly superior to Designs 1 and 2. However, the *CCI* of this design is lower than that of Design 3. But the

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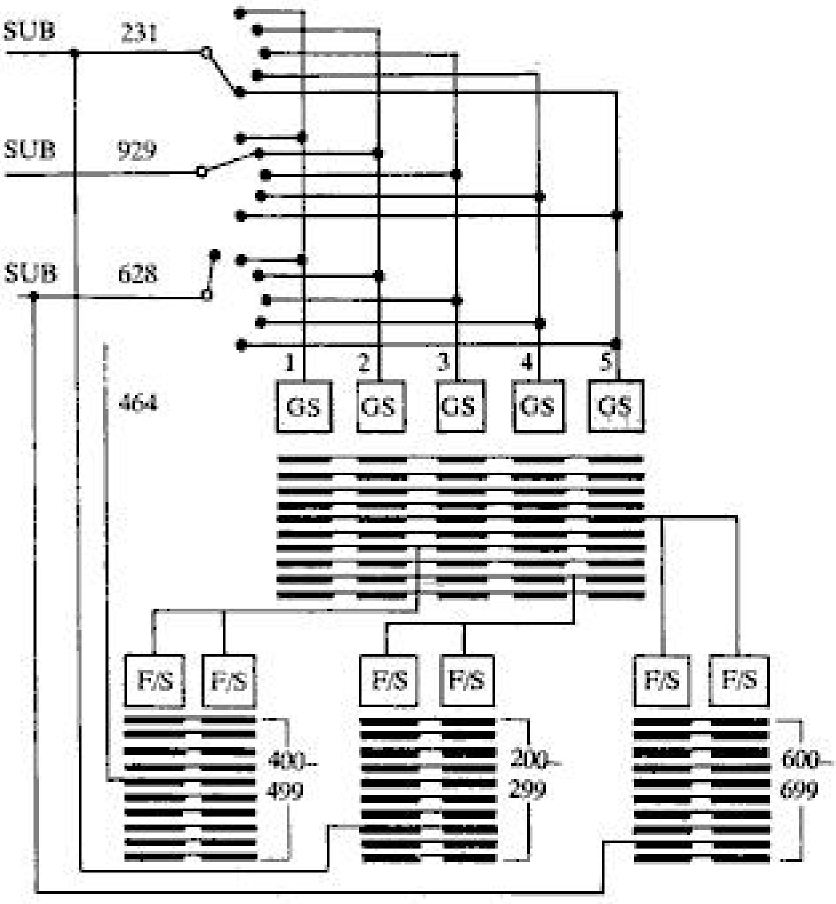
The trunking diagram for a 1000-line exchange is given in Fig. 2.13. As in the case of 100-line exchange, when a subscriber lifts his receiver, the pre-



F/S = final selector GS = group selector SUB = subscriber

Fig. 2.13 Trunking diagram of a 1000-line exchange.

selector hunts for a free group selector. When a free group selector is obtained, the subscriber is given the dial tone. When the subscriber dials the first digit, the group selector steps up in the vertical direction according to the digit dialled, and hunts for a free final selector in one of its 10 outlets. If a free selector is obtained, it responds to the next two digits and a connection is established, otherwise an engaged tone is sent out to the subscriber.

Each final selector, which is a two-motion selector, provides 100 outlets, and we need a minimum of 10 final selectors to connect 1000 subscribers. With 10 final selectors, only 10 simultaneous calls can be established, which

3. How can a suppressor cam be designed such that the interdigit gap precedes the dialling pulses?
4. Explain the working of the trigger dial mechanism. How is this superior to cam dial mechanism?
5. A busy tone does not imply that the called party is actually engaged in a conversation. Explain.
6. A regular long-distance caller disconnects one of the call attempts immediately on hearing the ringing tone with a remark that the call has landed on a wrong number. Can he be right? Why?
7. A long-distance dialler hears four different types of call-in-progress signals while establishing a call. What can he conclude?
8. In an English-speaking country, a long distance caller hears the voice announcement 'Lines in this route are busy. Please try after some time'.

   Is it possible for him to determine which segment is busy? Compare the situation in a multilingual country like India.
9. Describe the working of a rotary switch. Differentiate between forward acting and reverse acting types.
10. In a 100-line Strowger exchange using 100 two-motion selectors, show the trunking diagram when the subscriber 85 establishes a connection to subscriber 58. How does the diagram change if the call is initiated by subscriber 58?
11. Give the relative positions of the number pairs (61, 60) and (05, 35) in a two-motion selector.
12. What are the basic approaches to the design of subscriber access to Strowger systems? Describe them.
13. Describe how a uniselector can be used as a selector hunter or line finder.
14. Distinguish between earth testing and battery testing as applied to hunting operations in Strowger exchanges. Discuss the relative merits of each method.
15. A 1000-line exchange has 24 group selectors and 20 final selectors. How many simultaneous calls can be put through this exchange? How many simultaneous calls in the number range 200–299 can be put through if final selectors are uniformly distributed?
16. A 1000-line exchange has 24 group selectors and 50 final selectors uniformly distributed. How many simultaneous calls can be put through the exchange? How many simultaneous calls in the range 200 – 299 can be put through?

Let 1457 be the subscriber to be called in Exchange *F*. From Exchange *A*, the called subscriber can be reached by dialling either of the following number sequence:

For route *A-B-C-J-F*	01-04-03-01-1457
For route *A-I-H-G-F*	02-05-01-02-1457

The difficulties are now obvious:

- Identification number of a subscriber is route dependent.
- A user must have knowledge of the topology of the network and outlet assignments in each exchange.
- Depending on from which exchange the call originates, the number and its size vary for the same called subscriber.

These difficulties can be overcome if the routing is done by the exchange and a uniform numbering scheme is presented as far as the user is concerned. A number may now consist of two parts: An exchange identifier and a subscriber line identifier within the exchange. An exchange must have the capability of receiving and storing the digits dialled, translating the exchange identifier into routing digits, and transmitting the routing and the subscriber line identifier digits to the switching network. This function is performed by the Director subsystem in a Strowger exchange. Some important observations are in order with regard to the Director system:

- As soon as the translated digits are transmitted, the Director is free to process another call and is not involved in maintaining the circuit for the conversation.
- Call processing takes place independent of the switching network.
- A user is assigned a logical number which is independent of the physical line number used to establish a connection to him. The logical address is translated to actual physical address for connection establishment by an address translation mechanism.

All the above are fundamental features of a common control system. A functional block diagram of a common control switching system is shown in Fig. 3.2. The control functions in a switching system may be placed under four broad categories:

1. Event monitoring
2. Call processing
3. Charging
4. Operation and maintenance.

Events occurring outside the exchange at the line units, trunk junctors and interexchange signalling receiver/sender units are all monitored by the control subsystem. Typical events include call request and call release signals at the line units. The occurrences of the events are signalled by operating

is limited to 10 distinct signals, whereas a higher number would enhance signalling capability significantly. Finally, a more convenient method of signalling than rotary dialling is preferable from the point of view of human factors. These considerations led to the development of touch tone dial telephones in the 1950s, which were introduced first in 1964 after field trials. They are increasingly replacing rotary dial telephones all over the world.

The touch tone dialling scheme is shown in Fig. 3.3. The rotary dial is replaced by a push button keyboard. 'Touching' a button generates a 'tone'

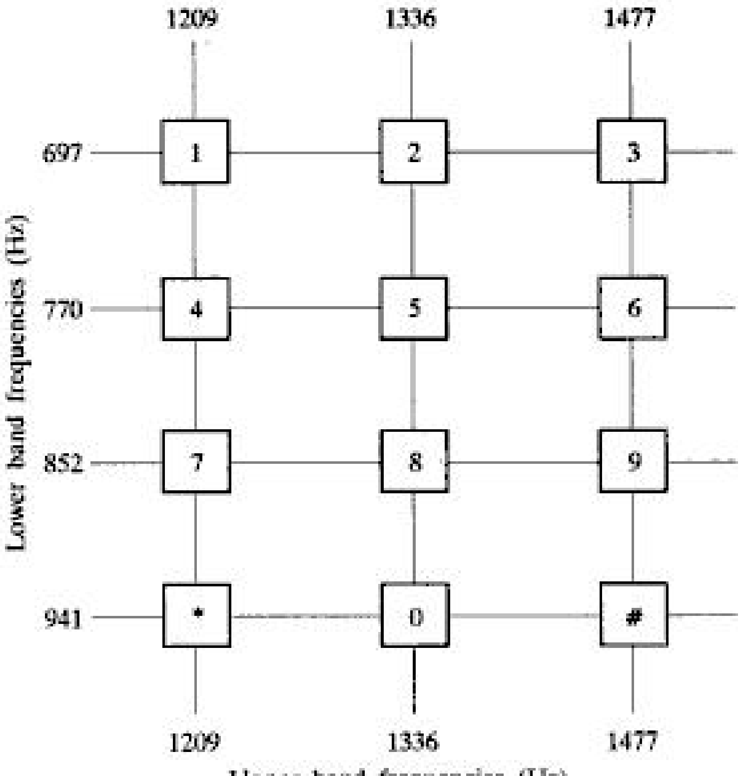


Fig. 3.3 Touch dial arrangement.

which is a combination of two frequencies, one from the lower band and the other from the upper band. For example, pressing the push button 9 transmits 852 Hz and 1477 Hz. An extended design provides for an additional frequency 1633 Hz in the upper band, and can produce 16 distinct signals. This design is used only in military and other special applications. Another design, known as **decadic push button** type, uses a push button dial in place

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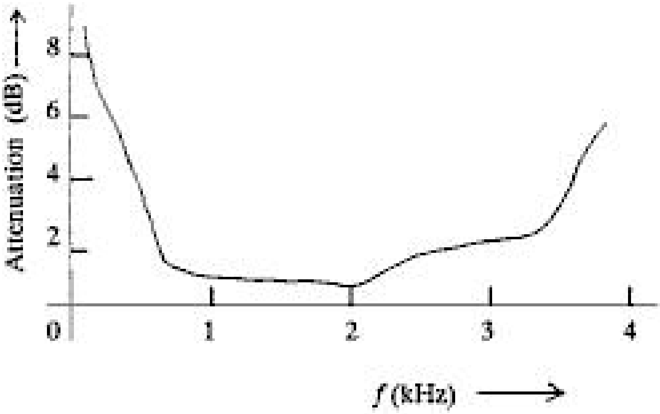
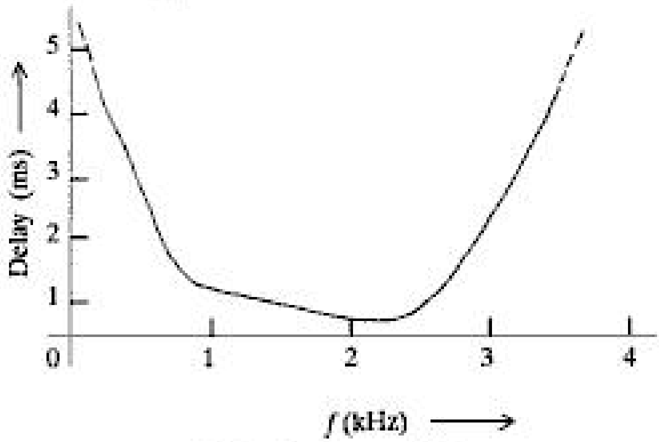


(a) Attenuation characteristics



(b) Delay characteristics

Fig. 3.5 Typical attenuation and delay characteristic of telephone networks.

relationships like 1:2 and 2:3 between adjacent two frequencies in the same band and between pairs of frequencies in the two different bands, respectively. Such a selection improves talk-off performance. As mentioned earlier, sounds composed of a multiplicity of frequencies at comparable levels are not likely to produce talk-off because of the limiter and selector design. Such sounds are produced by consonants. However, vowels are single frequency sounds with a series of harmonic components present in them. Susceptibility to talk-off due to vowels can be reduced by choosing the specific frequencies appropriately. The adjacent frequencies in the same band have a fixed ratio of 21:19, i.e, only the 21st and 19th harmonic

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connection *B-E*, the crosspoint *BC* may latch and *B* will be brought into the circuit of *A-C*. This is prevented by introducing an energising sequence for latching the crosspoints. *A* crosspoint latches only if the horizontal bar is energised first and then the vertical bar. (The sequence may well be that the vertical bar is energised first and then the horizontal bar). Hence the crosspoint *BC* will not latch even though the vertical bar *C* is energised as the proper sequence is not maintained. In order to establish the connection *B-E*, the vertical bar *E* needs to be energised after the horizontal bar is energised. In this case, the crosspoint *AE* may latch as the horizontal bar *A* has already been energised for establishing the connection *A-C*. This should also be avoided and is done by de-energising the horizontal bar *A* after the crosspoint is latched and making a suitable arrangement such that the latch is maintained even though the energisation in the horizontal direction is withdrawn. The crosspoint remains latched as long as the vertical bar *E* remains energised. As the horizontal bar *A* is de-energised immediately after the crosspoint *AC* is latched, the crosspoint *AE* does not latch when the vertical bar *E* is energised. Thus the procedure for establishing a connection in a crossbar switch may be summarised as:

energise horizontal bar		energise vertical bar
energise vertical bar	or	energise horizontal bar
de-energise horizontal bar		de-energise vertical bar

3.4 Crossbar Switch Configurations

In a nonblocking crossbar configuration, there are N^2 switching elements for N subscribers. When all the subscribers are engaged, only $N/2$ switches are actually used to establish connections. Table 3.1 shows the values of different design parameters (see Section 2.5) for four nonblocking switches. Unit cost is assumed for each crosspoint switching element. Providing N^2 crosspoints even for moderate number of users leads to impractical complex circuitry. A 1000-subscriber exchange would require 1 million crosspoint switches. Therefore, ways and means have to be found to reduce the number of switch contacts for a given number of subscribers.

Table 3.1 Nonblocking Crosspoint Switch Systems: Design Parameters

N	*S*	*SC*	*EUF*	*C*	*CCI*
4	16	2	12.50	16	0.5
16	256	8	3.13	256	0.5
64	4096	32	0.78	4096	0.5
128	16384	64	0.39	16384	0.5

N = No. of subscribers *S* = No. of switching elements
SC = switching capacity *C* = total cost
EUF = equipment utilisation factor *CCI* = cost capacity index

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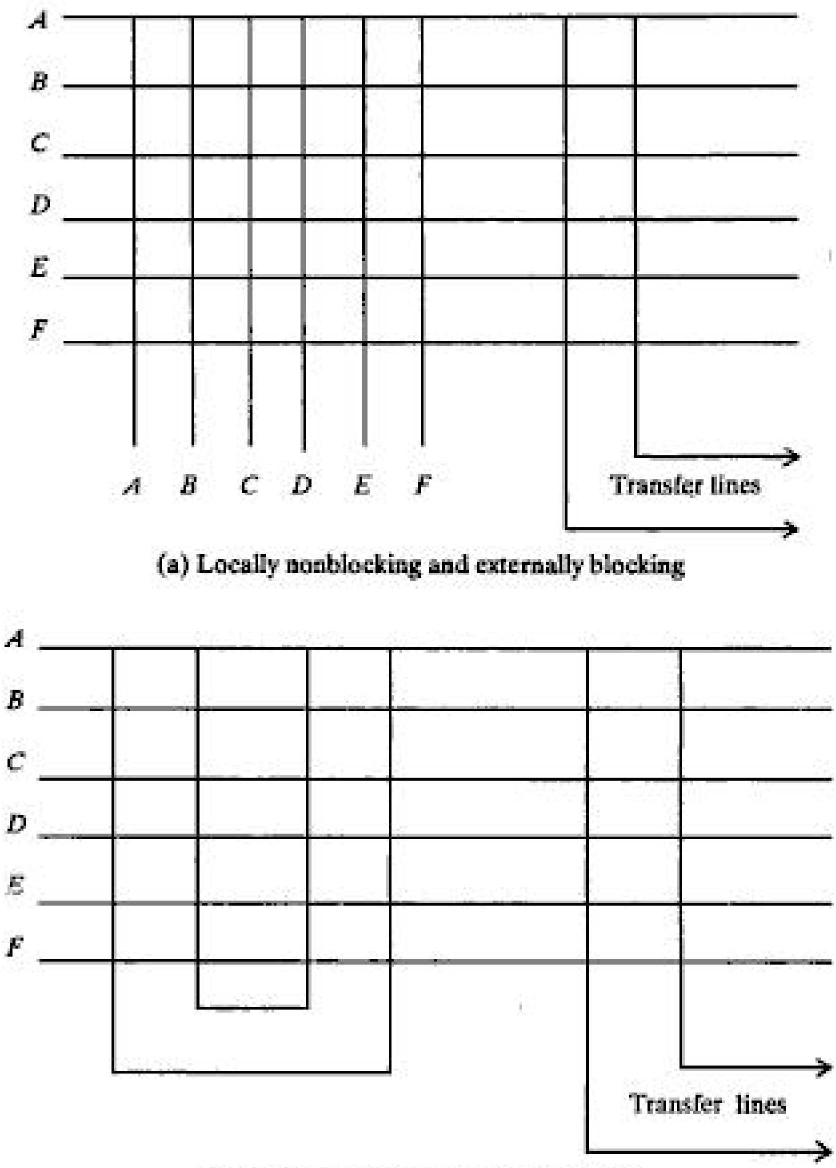


(a) Locally nonblocking and externally blocking

(b) Blocking both locally and externally

Fig. 3.10 Crossbar switches with transfer lines.

sealed in a glass tube as shown in Fig. 3.11. The sealing protects the electrical contacts from external contamination. The displacement involved in making contacts is about 0.2 mm. and this results in fast switching times

3. Talley, D., *Basic Telephone Switching Systems*, 2nd ed., Hayden Inc., New Jersey, 1979.

EXERCISES

1. 'Numbering plan in a telephone network must be independent of call routing'. Why? Explain.
2. What are the differences between common control and direct control?
3. List six events that may occur in a telephone system and the corresponding actions that may have to be taken by the common control system.
4. Calculate the time taken to dial a 12-digit number in a DTMF telephone when
   (a) the exchange is capable of receiving DTMF signals; and
   (b) the exchange can receive only pulse dialling.
   Compare the result with a rotary telephone dialling.
5. 'Contact bounce' can be a problem in DTMF telephone, i.e. a single press of a push button may be interpreted as more than one press. How does the DTMF dial design take this into account?
6. Show that the harmonic frequencies of any two adjacent base frequencies in DTMF telephone cannot match within the first 15 harmonics.
7. If the transmitted power of the low band frequency signal from a DTMF telephone is 1 mW, what should be the power in mW of the high band frequencies?
8. A telephone exchange supporting 5000 subscribers uses DTMF dialling and a common control subsystem with 100 digit receivers. Each digit receiver is assigned for a duration of five seconds per subscriber for call processing. If 20 per cent of the subscribers attempt to call simultaneously, what is the worst case wait time for a subscriber before he receives the dial tone?
9. A diagonal crosspoint matrix exchange supports 500 users. On an average 1000 calls are put through everyday. If the crosspoint contacts have a mean life of 10000 breaks and makes, estimate as to how often a crosspoint may be replaced in this exchange.
10. Estimate the number of crosspoints required to design an exchange that supports 500 users on a nonblocking basis and 50 transit, outgoing or incoming calls simultaneously.
11. Compare the reliabilities of one transistorised crosspoint switch and a bipolar chip containing 100 crosspoint switches. (Use known reliability data for the two technologies).

12. "The number of crossbars may be reduced by mounting contacts belonging to two subscribers on one bar". Can this be applied to both horizontal and vertical bars simultaneously? Explain how the scheme would work.

13. A blocking crossbar switch is to be designed to support 1000 subscribers. If the estimated peak traffic is 10 erlangs with average holding times of three minutes per call, estimate the number of crosspoints required.

4

Electronic Space Division Switching

Early crossbar systems were slow in call processing as they used electromechanical components for common control subsystems. Efforts to improve the speed of control and signalling between exchanges led to the application of electronics in the design of control and signalling subsystems. In late 1940s and early 1950s, a number of developmental efforts made use of vacuum tubes, transistors, gas diodes, magnetic drums and cathode ray tubes for realising control functions. Circuits using gas tubes were developed and employed for timing, ring translation and selective ringing of party lines. Vacuum tubes were used in single frequency signalling and transistors in line insulation test circuits. Contemporary to these developments was the arrival of modern electronic digital computers. Switching engineers soon realised that, in principle, the registers and translators of the common control systems could be replaced by a single digital computer.

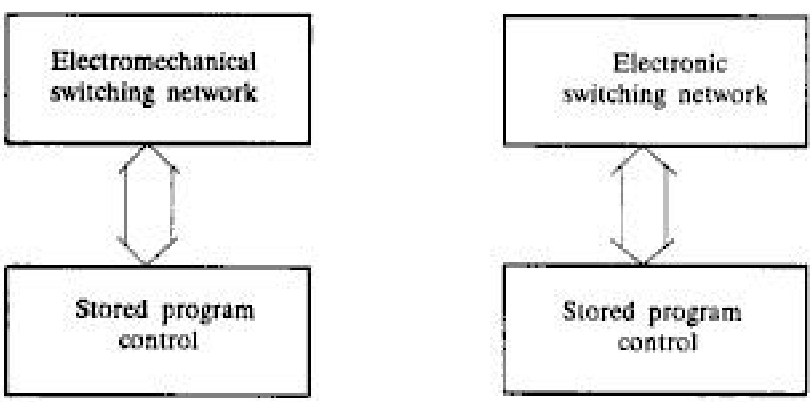
4.1 Stored Program Control

Modern digital computers use the stored program concept. Here, a program or a set of instructions to the computer is stored in its memory and the instructions are executed automatically one by one by the processor. Carrying out the exchange control functions through programs stored in the memory of a computer led to the nomenclature **stored program control** (SPC). An immediate consequence of program control is the full-scale automation of exchange functions and the introduction of a variety of new services to users. Common channel signalling (CCS), centralised maintenance and automatic fault diagnosis, and interactive human-machine interface are some of the features that have become possible due to the application of SPC to telephone switching.

Introducing a computer to carry out the control functions of a telephone exchange is not as simple as using a computer for scientific or commercial data processing. A telephone exchange must operate without interruption, 24 hours a day, 365 days a year and for say, 30–40 years. This means that the computer controlling the exchange must be highly tolerant to faults. Fault

tolerant features were unknown to early commercial computers and the switching engineers were faced with the task of developing fault tolerant hardware and software systems. In fact, major contributions to fault tolerant computing have come from the field of telecommunication switching.

Attempts to introduce electronics and computers in the control subsystem of an exchange were encouraging enough to spur the development of full-fledged electronic switching system, in which the switching network is also electronic. After about 10 years of developmental efforts and field trials, the world's first electronic switching system, known as No.1 ESS, was commissioned by AT&T at Succasunna, New Jersey, in May 1965. Since then, the history of electronic switching system and stored program control has been one of rapid and continuous growth in versatility and range of services. Today, SPC is a standard feature in all the electronic exchanges. However, attempts to replace the space division electromechanical switching matrices by semiconductor crosspoint matrices have not been greatly successful, particularly in large exchanges, and the switching engineers have been forced to return to electromechanical miniature crossbars and reed relays, but with a complete electronic environment. As a result, many space division electronic switching systems use electromechanical switching networks with SPC. Nonetheless, private automatic branch exchanges (PABX) and smaller exchanges do use electronic switching devices. The two types of space division electronic switching systems, one using electromechanical switching network and the other using electronic switching network, are depicted in Fig. 4.1. Both the types qualify as electronic switching systems although only one of them is fully electronic. With the evolution of time division switching, which is done in the electronic domain, modern exchanges are fully electronic. Principles of time division switched electronic exchanges are discussed in Chapter 6.



(a) Electromechanical switching (b) Electronic switching

Fig. 4.1 Electronic space division switching systems.

There are basically two approaches to organising stored program control: centralised and distributed. Early electronic switching systems (ESS) developed during the period 1970–75 almost invariably used centralised control. Although many present day exchange designs continue to use centralised SPC, with the advent of low cost powerful microprocessors and very large scale integration (VLSI) chips such as programmable logic arrays (PLA) and programmable logic controllers (PLC), distributed SPC is gaining popularity.

4.2 Centralised SPC

In centralised control, all the control equipment is replaced by a single processor which must be quite powerful. It must be capable of processing 10 to 100 calls per second, depending on the load on the system, and simultaneously performing many other ancillary tasks. A typical control configuration of an ESS using centralised SPC is shown in Fig. 4.2. A centralised SPC

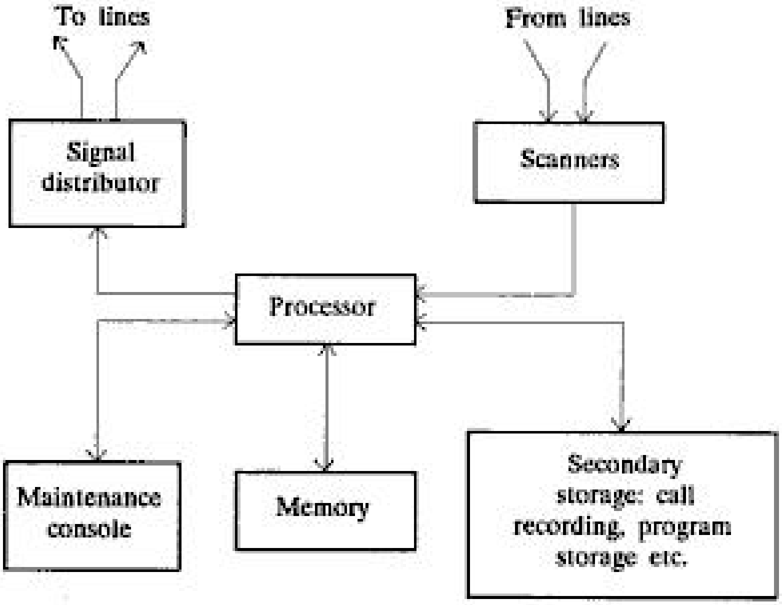
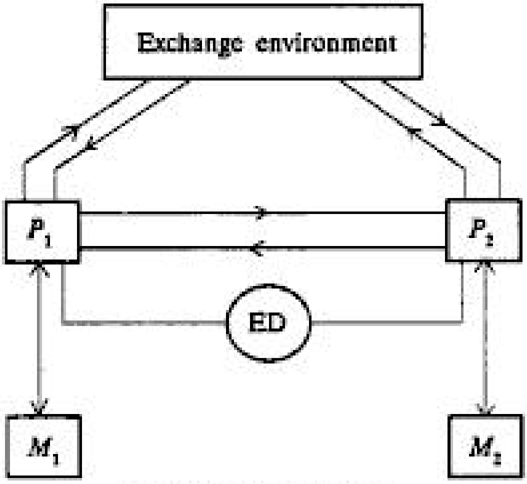


Fig. 4.2 Typical centralised SPC organisation.

configuration may use more than one processor for redundancy purposes. Each processor has access to all the exchange resources like scanners and distribution points and is capable of executing all the control functions. A redundant centralised structure is shown in Fig. 4.3. Redundancy may also be provided at the level of exchange resources and function programs. In actual implementation, the exchange resources and the memory modules contain-



ED = exclusion device

Fig. 4.6 Load sharing configuration.

processors exchange information needed for mutual coordination and verifying the 'state of health' of the other. If the exchange of information fails, one of the processors which detects the same takes over the entire load including the calls that are already set up by the failing processor. However, the calls that were being established by the failing processor are usually lost. Sharing of resources calls for an exclusion mechanism so that both the processors do not seek the same resource at the same time. The mechanism may be implemented in software or hardware or both. Figure 4.6 shows a hardware exclusion device which, when set by one of the processors, prohibits access to a particular resource by the other processor until it is reset by the first processor. Software exclusion mechanism is discussed in detail in Section 4.4.

Under normal operation, each processor handles one-half of the calls on a statistical basis. The exchange operators can, however, send commands to split the traffic unevenly between the two processors. This may be done, for example, to test a software modification on one processor at low traffic, while the other handles majority of the calls. Load sharing configuration gives much better performance in the presence of traffic overloads as compared to other operating modes, since the capacities of both the processors are available to handle overloads. Load sharing configuration increases the effective traffic capacity by about 30 per cent when compared to synchronous duplex operation. Load sharing is a step towards distributed control.

One of the main purposes of redundant configuration is to increase the overall availability of the system. A telephone exchange must show more or less a continuous availability over a period of perhaps 30 or 40 years. We now

4.3 Distributed SPC

In distributed control, the control functions are shared by many processors within the exchange itself. This type of structure owes its existence to the low cost microprocessors. This structure offers better availability and reliability than the centralised SPC.

Exchange control functions may be decomposed either 'horizontally' or 'vertically' for distributed processing. In vertical decomposition, the exchange environment is divided into several blocks and each block is assigned to a processor that performs all control functions related to that block of equipments. The total control system now consists of several control units coupled together. The processor in each block may be duplicated for redundancy purposes and operates in one of the three dual processor operating modes discussed in Section 4.2. This arrangement is modular so that the control units may be added to handle additional lines as the exchange is expanded.

In horizontal decomposition, each processor performs only one or some of the exchange control functions. A typical horizontal decomposition is along the lines of the functional groupings shown in Fig. 4.7. A chain of different processors may be used to perform the event monitoring, call processing and O&M functions. The entire chain may be duplicated as illustrated in Fig. 4.9 for providing redundancy. Similar operating principles as in the case of dual processor structure apply to the dual chain configuration.

4.3.1 Level 3 Processing

Since the processors perform specific functions in distributed control, they can be specially designed to carry out these functions efficiently. In Fig. 4.9, level 3 processor handles scanning, distribution and marking functions. The processor and the associated devices are located physically close to the switching network, junctors and signalling equipment. Processing operations involved are of simple, specialised and well-defined nature. Generally, processing at this level results in the setting or sensing of one or more binary conditions in flipflops or registers. It may be necessary to sense and alter a set of binary conditions in a predefined sequence to accomplish a control function. Such simple operations are efficiently performed either by wired logic or microprogrammed devices.

A control unit, designed as a collection of logic circuits using logic elements, electronic or otherwise, is called a 'hard-wired' control unit. A hard-wired unit can be exactly tailored to the job in-hand, both in terms of the function and the necessary processing capacity. But it lacks flexibility and cannot be easily adapted to new requirements. A microprogrammed unit is more universal and can be put to many different uses by simply modifying the microprogram and the associated data. With the same technology, the microprogrammed units tend to be more expensive and slower than hard-wired

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6. It provides unambiguous specifications and descriptions for tendering and evaluation of offers.
7. It provides a basis for meaningful comparison of the capabilities of different SPC systems.

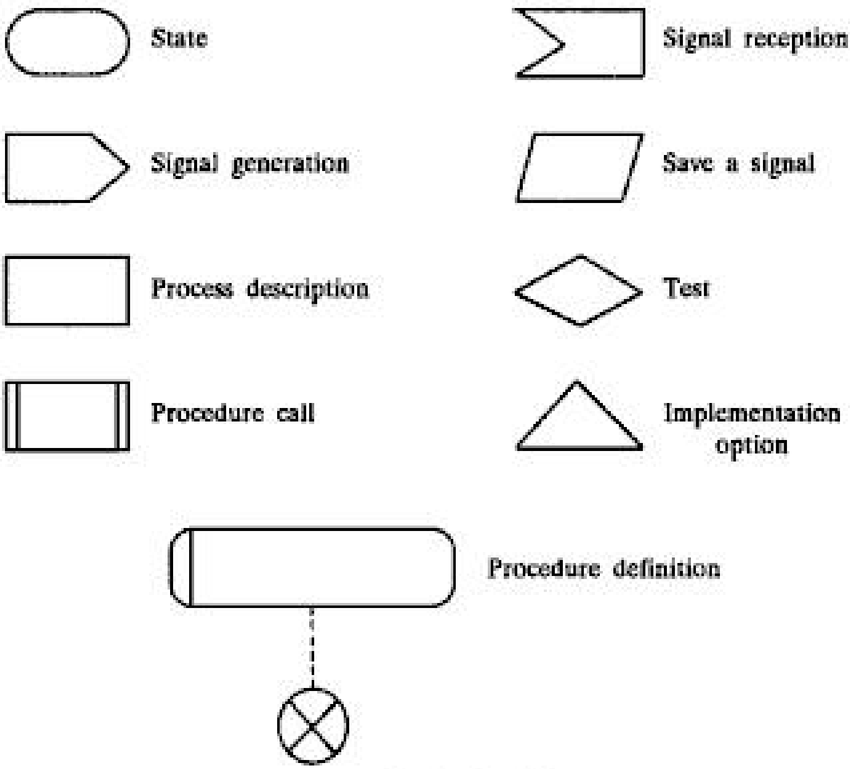

Fig. 4.13 Standard symbols in SDL.

Switching systems basically belong to the class of finite state machines (FSM) which are asynchronous in nature and follow a sequential logic for their operation. They can be modelled by using a combinational part and a memory part as shown in Fig. 4.14. In FSM, the status of the output circuits not only depends upon the inputs but also upon the current state of the machine. Asynchronous sequential operation gives rise to many problems due to transient variations that may occur in the logic circuits and memory elements. Clocked synchronous operation shown in Fig. 4.15 overcomes such problems. The theory of the operating principles of synchronous finite state machines forms the basis of design of SDL.

Both assembly and high level languages are used in producing switching software. Early electronic switching systems used assembly language programming extensively. The present trend is to use more and more of high

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4.5 Application Software

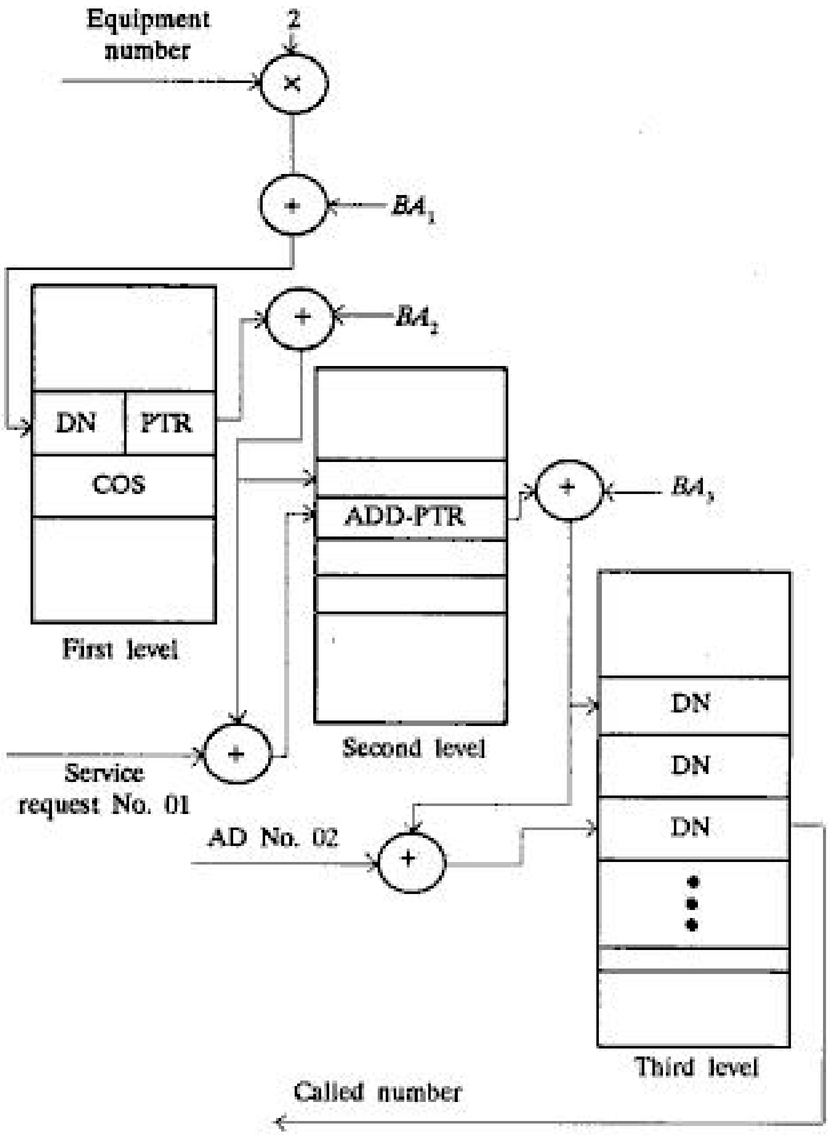
The application software of a switching system may be divided into three main classes:

1. Call processing software
2. Administrative software
3. Maintenance software.

A software package is described by its organisation, the data structures it uses and the processing functions it performs. Application software packages of a switching system use a modular organisation. The software packages are divided into program modules, each dealing with a specific task. The size of a module varies depending on the task. Generally speaking, the modules are not self-contained. They exchange data with other modules, either directly through interfaces or indirectly through data tables. Several modules are grouped together to constitute functional units corresponding to independent functions. A module may be a part of more than one function unit. Usually, a functional unit runs as a separate process in the system. The modules of a process are strung together through special programs or chaining tables. Module chaining through tables is illustrated in Fig. 4.16. Associated with every module is a pointer to a set of entries in the chaining table pertaining to that module. Each entry in the chaining table consists of a key and a module number. Whenever a module completes execution, its corresponding entries in the chaining table are scanned and the keys are compared to a function status key. If a match occurs, the corresponding module in the chaining entry is executed next. This approach provides flexibility for adding new modules to a function or deleting old modules by simply modifying the chaining data.

Application software accounts for about 80 per cent of the total volume of the software in a switching system. Administration and maintenance programs together constitute about 65 per cent of the total volume. The total software typically comprises between 400,000 and 500,000 machine instructions. The entire software need not be **core resident**. Considering the real time constraints, the system software and the call processing application software are usually core resident. The administration and maintenance modules reside on a back up storage and are brought into the main memory as and when required. Depending on the architectural support available from the switching processor, the operating system may use overlay or **virtual memory** technique for this purpose.

Switching system software almost always uses a parameterised design. This enables the same package to be used over a wide range of exchanges by adapting the package to specific exchange characteristics. The parameters may be divided into system parameters and office parameters. The system parameters afford flexibility at the overall system level while the office parameters define program execution limits at specific exchanges. The system



AD = abbreviated dialling ADD = abbreviated dial directory
BA = base address COS = class of service
DN = directory number PTR = pointer

Fig. 4.17 Access to calling line data.

to the calling subscriber is obtained. The user request for a service is converted to an offset which is added to the starting address of the entry to

flow, uncover traffic sensitive network or terminal problems and gather information for billing. If the traffic load exceeds the capacity of the system, an overload control process is initiated which reschedules priorities and frequencies of activities to ensure that the system continues to process as many calls as practicable. One way of overload control is to restrict the number of call originations per unit time. This is done by delaying the sending of dial tone for a few seconds to a subscriber who goes off-hook.

Maintenance programs are run for performing either diagnostic function or preventive maintenance. During periods of normal traffic, there are preventive maintenance programs that take advantage of unused real time to run test programs of hardware and to audit system memory contents for correctness and consistency. In periods of high traffic, these programs are deferred. If a fault occurs in the system, the operating system activates unscheduled maintenance programs to recover the system from the fault with minimal mutilation of calls in progress. Sections of the exchange hardware may be isolated and diagnostic program run to enable maintenance personnel to fix faults.

4.6 Enhanced Services

One of the immediate benefits of stored program control is that a host of new or improved services can be made available to the subscribers. Over a hundred new services have already been listed by different agencies like CCITT, and the list is growing day by day. In fact, the only limitations in introducing new services seem to be the imagination of the designers and the price the market is prepared to pay for the services. Although there are a large number of services, they may be grouped under four broad categories:

1. Services associated with the calling subscriber and designed to reduce the time spent on dialling and the number of dialling errors
2. Services associated with the called subscriber and designed to increase the call completion rate
3. Services involving more than two parties
4. Miscellaneous services.

These new services are known as supplementary services and some of the prominent ones are as follows:

Category 1:

- Abbreviated dialling
- Recorded number calls or no dialling calls
- Call back when free.

Category 2:

- Call forwarding

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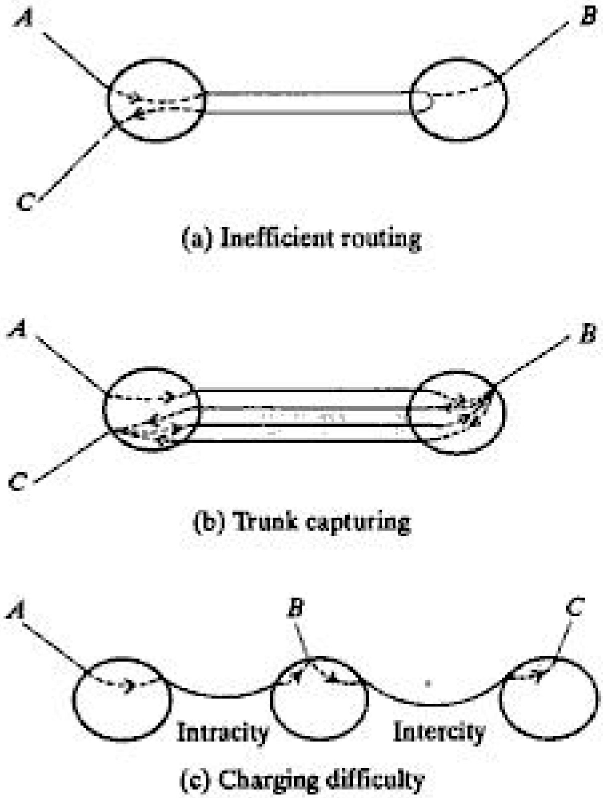

Fig. 4.20 Difficulties in call forwarding across exchanges.

Call waiting feature provides an indication to a busy subscriber that another party is trying to reach him. The indication is given through a short audible tone, lasting typically about three seconds. The subscriber may then

- ignore the incoming call and continue with the present one,
- place the incoming call on hold and continue with the first call,
- place the first call on hold and answer the new call, or
- release the first call and accept the new one.

Call-waiting feature requires two switching paths to be set up simultaneously. Both the paths must use the same signalling scheme.

Consultation hold is a facility that enables a subscriber in conversation to place the other subscriber on hold and contact a third subscriber for consultation. This is like the telephone extension service used in offices where a secretary may consult the executive while holding an incoming call except that any subscriber number can be dialled for consultation. It may be

K inputs of the second stage can be connected to any of the N outlets. As a result, there are K alternative paths for any inlet/outlet pair connection. The network is said to provide **full connectivity** or **full availability**, in the sense that any of the N inlets can be connected to any of the N outlets in the network. The term *full connectivity* must be distinguished from the term *fully connected network* defined in Section 1.1. Each stage of the network has NK switching elements. Assuming about 10 per cent of the subscribers to be active on an average, K may be set equal to $(N/16)$. In this case, the number of switching elements, S, in the network is $(N^2/8)$. For $N = 1024$, we have $K = 64, S = 131{,}072$.

For large N, the switching matrix $N \times K$ may still be difficult to realise practically. It is necessary to consider architectures that use smaller sized switching matrices. Let us consider the two-stage realisation of an $M \times N$ switch using a number of smaller switching matrices as shown in Fig. 4.23.

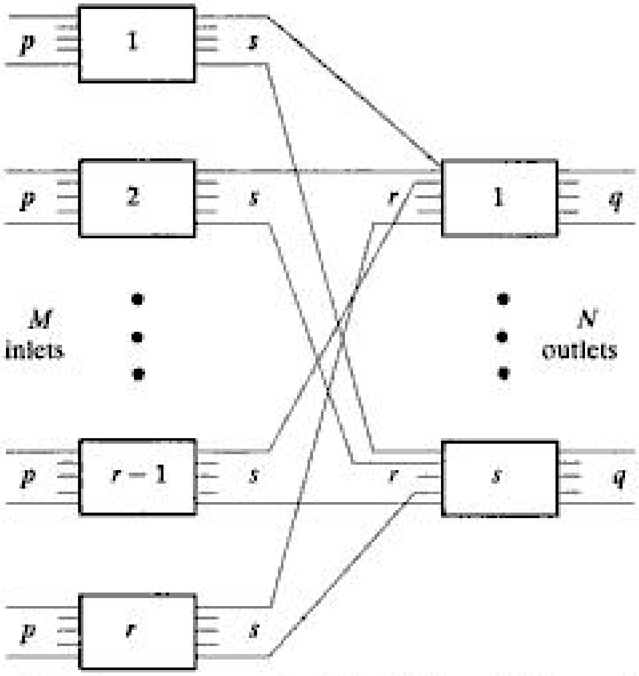

Fig. 4.23 Two-stage network with multiple switching matrices in each stage.

M inlets are divided into r blocks of p inlets each such that $M = pr$. Similarly, the N outlets are divided into s blocks of q outlets each such that $N = qs$. In order to ensure full availability, there must be at least one outlet from each block in the first stage terminating as inlet on every block of the second stage.

stage 3. Unlike the two-stage network discussed in Section 4.7, here any arbitrary inlet in the first stage has s alternative paths to reach any arbitrary outlet in the third stage. The total number of switching elements is given by

$$S = rps + sr^2 + spr = 2Ns + sr^2 = s(2N + r^2) \tag{4.14}$$

If we use square matrices in the first and third stages, we have $p = s = (N/r)$ and, therefore,

$$S = \frac{2N^2}{r} + Nr \tag{4.15}$$

Equation (4.15) indicates that there is an optimum value for r that would minimise the value of S. To obtain this value of r, we differentiate Eq. (4.15), set it equal to zero and determine the value of r :

$$\frac{dS}{dr} = \frac{-2N^2}{r^2} + N = 0$$

Therefore, $r = \sqrt{2N}$. The second derivative, being positive at this value of r, indicates that the value of S is minimum, i.e.

$$S_{\min} = 2N\sqrt{2N} \tag{4.16}$$

and $p = N/r = \sqrt{N/2}$. The optimum ratio of the number of blocks to the number of inputs per block is given by

$$r/p = \sqrt{2N}/\sqrt{N/2} = 2 \tag{4.17}$$

There are a variety of techniques that can be used to evaluate the blocking probabilities of multistage switching networks. Of these, two are widely used: one due to C.Y. Lee and the other due to C. Jacobaeus (see Further Reading). Both the techniques are approximate techniques and provide reasonably accurate results, particularly when comparisons of alternative structures are more important than absolute numbers. The model proposed by Jacobaeus is somewhat more accurate than the one proposed by Lee. But the greatest value of Lee's approach is in the ease of modelling and the fact that the model and the associated formulae directly relate to the underlying network structures. In this book, we use Lee's probability graphs to estimate the blocking probability of multistage networks.

A probability graph of a three-stage network is shown in Fig. 4.25. In the graph, the small circles represent the switching stages and the lines represent the interstage links. The network graph shows all possible paths between a given inlet and an outlet. The graph reflects the fact that there are s alternative paths for any particular connection, one through each block in the second stage.

Blocking probabilities in the network may be estimated by breaking down a graph into serial and parallel paths. Let

Table 4.4 Switch Advantage Ratio in Three-Stage Networks

N	Number of switching elements, S		λ
	Single stage	Three stages	
128	16,384	8,192	2
2,048	4 million	0.5 million	8
8,192	64 million	4 million	16
32,768	1 billion	32 million	32

As the value of N increases, we get relatively better savings in the number of switching elements. But the actual number of switching elements still becomes impracticably large for large values of N. For example, a 30,000-line nonblocking exchange needs about 30 million switching elements.

The number is unmanageable even in blocking exchanges. Further reductions in the number of switching elements are possible by using even higher number of stages than three.

4.9 n-Stage Networks

A variety of ways exist in which switching networks with four or more stages can be constructed. A description of all such networks is beyond the scope of this book. As an illustrative example, we discuss a five-stage network shown in Fig. 4.27. This network is formed by replacing each block of the centre stage of the network shown in Fig. 4.24 with a three-stage network. There are r inlets to a block in the centre stage of the network in Fig. 4.24. These are now terminated on the three-stage network in Fig. 4.26 that replaces the block in Fig. 4.24. The r inlets are distributed among the r_1 blocks shown in Fig. 4.26 with (r/r_1) inlets per block.

In order to compare the requirements of the switching elements in the case of three-stage and five-stage networks, let us assume that the three-stage network is realised with optimum number of square blocks in each stage so that the minimum number of switching elements are used. Taking a specific example of 2^{15} subscribers, for the three-stage network, from Eq.(4.16) we have the relations

$$S = 16 \times 2^{20}, \qquad p = 128, \qquad r = 256$$

In order to maintain the same level of blocking performance for the five-stage network as the three-stage network, let us assume that the centre three stages of the five-stage network are designed to be nonblocking and estimate the number of switching elements:

Switching element in the first stage $= 2^8 \times 2^7 \times 2^7 = 2^{22}$
Switching elements in the last stage $= 2^{22}$

15. Determine the design parameters of a three-stage switch with inlet utilisation of 0.1 to achieve a $P_B = 0.002$ for (a) $N = 128$, (b) $N = 2048$, and (c) $N = 8192$.

16. Using the Lee graph, show that the blocking probability of a five-stage network is given by

$$P_B = [1 - (1 - \alpha_1)^2 [1 - \{1 - (1 - \alpha_2^2) s_1\}]^s$$

where $\alpha_1 = \alpha(p/s)$, $\alpha_2 = \alpha_1 \dfrac{r}{r_1 s_1}$, α is the probability that an input line is active and r, r_1 and s_1 have the same significance as in Fig.4.27.

0–H Hz to pass. If f_s is less than twice H, portions of PAM signal spectrum will overlap as shown in Fig. 5.1(c). This overlapping of the sidebands produces beat frequencies that interfere with the desired signal and such an interference is referred to as **aliasing or foldover distortion**. It may be noted that aliasing is a phenomenon common to all sample data systems. As mentioned earlier, toll quality speech is band limited to 300–3400 Hz to conserve bandwidth while maintaining intelligibility. To digitise this wave-form, the minimum sampling frequency required is 6.8 kHz to avoid aliasing effects. In digital telephone networks, speech is sampled at 8 kHz rate. In this context, the filter used for band limiting the input speech waveform is known as **antialiasing filter**. It may be noted that for a 3.4 kHz cut-off frequency, 8 kHz sampling results in oversampling. This oversampling provides for the nonideal filter characteristics such as lack of sharp cut off. The sampled signal is sufficiently attenuated at the overlap frequency of 4 kHz to adequately reduce the energy level of the foldover spectrum.

5.2 Quantisation and Binary Coding

Pulse amplitude modulation systems are not generally useful over long distances, owing to vulnerability of the individual pulse amplitudes to noise, distortion and crosstalk. The amplitude susceptibility may be reduced or eliminated by converting the PAM samples into a digital format, thereby allowing the use of regenerative repeaters to remove transmission imperfections before errors result. A finite number of bits are used for coding PAM samples. With n bits, the number of sample values that can be represented is 2^n. But the PAM sample amplitudes can take on an infinite range of values. It, therefore, becomes necessary to quantise the PAM sample amplitude to the nearest of a range of discrete amplitude levels.

The process of quantisation is depicted in Fig. 5.2. Signal V is confined to a range from V_L to V_H, and this range is divided into M ($M = 8$ in Fig. 5.2) equal steps. The step size S is given by

$$S = (V_H - V_L)/M \tag{5.2}$$

In the centre of each of these steps we locate the quantisation levels $V_0, V_1, \ldots, V_{M-1}$. The quantised signal V_q takes on any one of the quantised level values. A signal V is quantised to its nearest quantisation level. The boundary values between the steps are equidistant from two quantisation levels and a convention may be adopted to quantise them to one of the levels. For example,

$$V_q = V_3 \quad \text{if } (V_3 - S/2) \le V < (V_3 + S/2)$$
$$V_q = V_4 \quad \text{if } (V_4 - S/2) \le V < (V_4 + S/2)$$

Thus, the signal V_q makes a quantum jump of step size S and at any instant of time the quantisation error $V - V_q$ has a magnitude which is equal to or less

$$\sigma^2 = \int_{-S/2}^{S/2} (e-0)^2 \frac{1}{S}\, de$$

$$= \frac{1}{S} \int_{-S/2}^{S/2} e^2\, de$$

$$= \frac{1}{S}\left(\frac{e^3}{3}\right)_{-S/2}^{S/2} = \frac{S^2}{12} \qquad (5.3b)$$

Signal to quantisation noise ratio (SQR) is a good measure of performance of a PCM system transmitting speech. If V_r is the r.m.s. value of the input

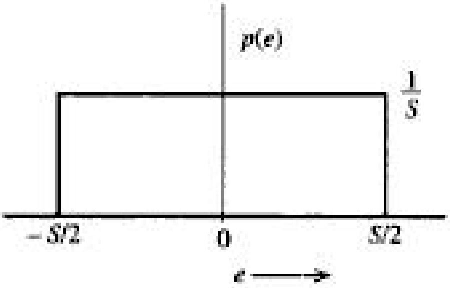

Fig. 5.5 Probability distribution of error due to linear quantisation.

signal and if we assume (for convenience) a resistance level of 1 ohm, then SQR is given by

$$\begin{aligned} \mathrm{SQR} &= 10 \log [(V_r^2)/(S^2/12)] \text{ dB} \\ &= 10 \log (12) + 20 \log (V_r/S) \text{ dB} \\ &= 10.8 + 20 \log (V_r/S) \text{ dB} \end{aligned} \qquad (5.4)$$

If the input signal is a sinusoidal wave with V_m as the maximum amplitude, *SQR* may be calculated for the full range sine wave as

$$\begin{aligned} \mathrm{SQR} &= 10 \log [(V_m/\sqrt{2})^2)/(S^2/12)] \text{ dB} \\ &= 10 \log (6) + 20 \log (V_m/S) \text{ dB} \\ &= 7.78 + 20 \log (V_m/S) \text{ dB} \end{aligned} \qquad (5.5)$$

Expressing S in terms of V_m and the number of steps, M, we have

$$\begin{aligned} \mathrm{SQR} &= 10 \log \frac{(V_m^2/2)}{(4V_m^2/12\,M^2)} \text{ dB} \\ &= 10 \log (1.5\,M^2) \text{ dB} \\ &= 20 \log (1.225\,M) \text{ dB} \end{aligned} \qquad (5.6)$$

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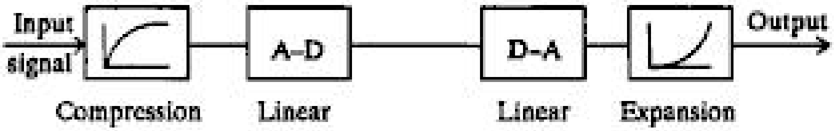

Fig. 5.10 The process of companding.

The input signal is first compressed by using a nonlinear functional device and then a linear quantiser is used. At the receiving end, the quantised signal is expanded by a nonuniform device having an inverse characteristic of the compression at the sending end. The process of first compressing and then expanding is referred to as **companding**.

A variety of nonlinear compression-expansion functions can be chosen to implement a compandor. The obvious one is a logarithmic law. Unfortunately, as is seen from Fig. 5.11, the function $y = \ln x$ does not pass

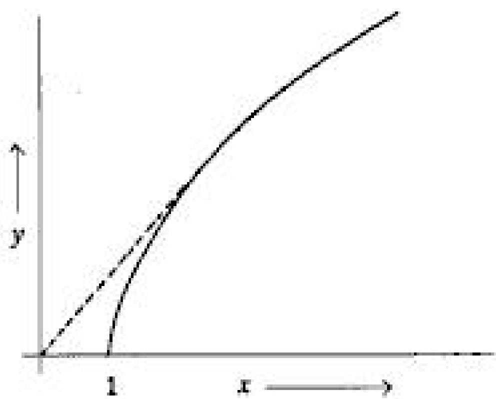

Fig. 5.11 Function $y = \ln x$.

through the origin. It is therefore necessary to substitute a linear portion to the curve for lower values of x. Most of the practical companding systems are based on a law suggested by K.W. Cattermole. For logarithmic section the law is

$$y = \frac{1 + \ln Ax}{1 + \ln A} \quad \text{for } \frac{1}{A} \le x \le 1 \tag{5.9}$$

and for linear section,

$$y = \frac{Ax}{1 + \ln A} \quad \text{for } 0 \le x \le \frac{1}{A} \tag{5.10}$$

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- Nonuniform amplitude distributions
- Sample-to-sample correlations
- Periodicity or cycle-to-cycle correlations
- Pitch interval-to-pitch interval correlations
- Speech pauses or inactivity factors.

In speech signal, lower amplitude values are more common than the higher amplitude values. The power levels of active speech signals tend to occur at the lower end of the signal range. There is a strong correlation between adjacent 8 kHz samples and the correlation coefficient is estimated to be 0.85 or higher. Although telephone quality speech signal requires the entire 300 – 3400 Hz bandwidth, at any instant of time only a few frequencies exist in a sound and the waveform tends to exhibit strong correlations over several samples corresponding to cycles of an oscillation. A sizeable fraction of the human speech sounds is produced by the flow of puffs of air from the lungs into the vocal tract. The interval between these puffs of air is known as the pitch interval. Speech waveforms display repetitive patterns corresponding to the duration of a pitch interval in a given sound. There may be as many as 20 to 40 pitch intervals in a single sound. Analysis of telephone conversations have indicated that a party is typically active for about 40% of a call duration largely on account of listening to the other party's talk. In addition, there are pauses in one's speech. All these properties of speech are useful in designing coding schemes that reduce the bandwidth requirements of the channel. In the digital domain significant reductions in bit rates can be achieved. A number of coding schemes exploiting different redundancy properties have been evolved and we discuss some of these in this section and in Section 5.6.

Delta or differential coding systems are designed to take advantage of the sample-to-sample redundancies in speech waveforms. Because of the strong correlation between adjacent speech samples, large abrupt changes in levels do not occur frequently in speech waveforms. In such situations, it is more efficient to transmit or encode and transmit only the signal changes instead of the absolute value of the samples. **Delta modulation (DM)** is a scheme that transmits only the signal changes and **differential pulse code modulation (DPCM)** encodes the differences and transmits them.

A delta modulator may be implemented by simply comparing each new signal sample with the previous sample and transmitting the resulting difference signal. At the receiver end, the difference signals are added up to construct the absolute signal by using an integrator. However, such a system, being open loop, suffers from the possibility of the receiver output diverging from the transmitter input due to system errors or inaccuracies. The system can be converted into a closed loop system by setting up a feedback path with an integrator at the transmitting end as shown in Fig. 5.14. The integrator must be identical to the one at the receiving end. The difference signal $e(b)$ has only two possible levels. A positive level indicates that the new sample

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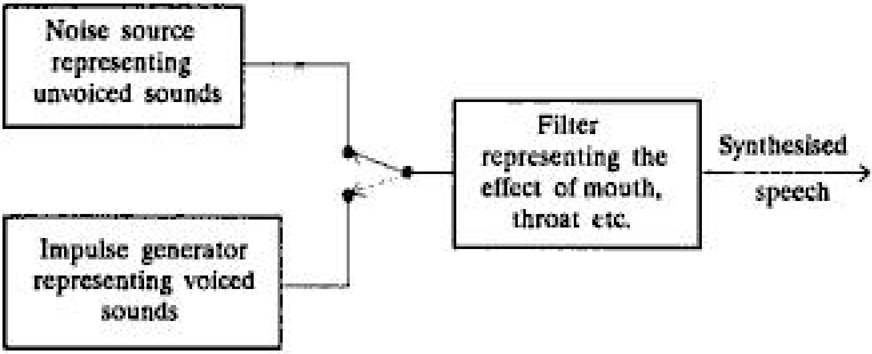

Fig. 5.16 Speech model used in vocoder design.

Fig. 5.16) for the required duration. The filter at the receiver end in Fig. 5.16 implements a vocal tract transfer function. It is constructed by using a bank of bandpass filters whose input power levels are the respective sub-band power levels at the transmitting end. Superposing the individual band outputs along with the appropriate switching of the signal source, generates, in a spectral sense, the original signal. Recent advances in digital signal processing (DSP) permit the use of Fourier transform algorithms to determine the input spectrum in lieu of the bank of analog filters.

The three or four energy peaks in the short-term spectral density of speech are known as formants. A formant vocoder determines the location and amplitude of these spectral peaks and transmits this information instead of the entire spectrum envelope.

In addition to determining the nature of excitation (voiced or unvoiced) and the pitch interval or period, the linear predictive coders (LPC) also predict parameters for vocal tract. The vocal tract filter (see Fig. 5.16) is an adjustable one whose parameters are varied based on the prediction. The LPC functions as a feedback system similar to adaptive DM and ADPCM. As a result, quality of speech synthesised by LPC is superior to that of the speech synthesised by the channel or formant vocoders.

5.7 Pulse Transmission

In the earlier sections of this chapter, we have been considering how analog signal waveforms are represented in the form of binary digital signals or pulses. In this section, we consider aspects relating to the communication of such pulse trains in practical transmission channels.

Any reasonably behaved periodic function $g(t)$ with period T can be represented in the form of a Fourier series:

$$g(t) = C/2 + \sum_{n=1}^{\infty} a_n \sin(2\pi nft) + \sum_{n=1}^{\infty} b_n \cos(2\pi nft) \qquad (5.13)$$

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Otherwise, the harmonics will not combine in proper amplitude and phase at the receiver with the result that the reconstructed waveform will be distorted. Measurable transmission line parameters corresponding to these two preceding requirements are **attenuation constant** A_c and **phase velocity** V_p. Another source of distortion is the reflection of the transmitted wave by the transmission line. The related parameter for this is the **characteristic impedance** Z_0 of the line.

The phase velocity V_p is a measure of the speed of wave propagation along the transmission line in meters per second. Ideally, it should be constant for all frequencies to avoid phase distortion. When it is not, certain frequencies may be delayed so much that they interfere with frequencies corresponding to later pulses. This is known as **intersymbol interference**. The attenuation constant A_c is a measure of the loss of the signal power per meter of the line length. A constant A_c value for all frequencies would avoid amplitude distortion. The characteristic impedance Z_0 is the inherent impedance that is presented to the signal by the transmission line. If the source and load impedances are equal to the characteristic impedance, there is no reflection of the wave from the load, and hence there is no distortion.

In a transmission line, resistance, inductance and capacitance are distributed along the entire length of the line. For purposes of analysis, the distributed parameters are lumped usually over a distance of one meter and an equivalent circuit formed as shown in Fig. 5.18. R_s represents the series

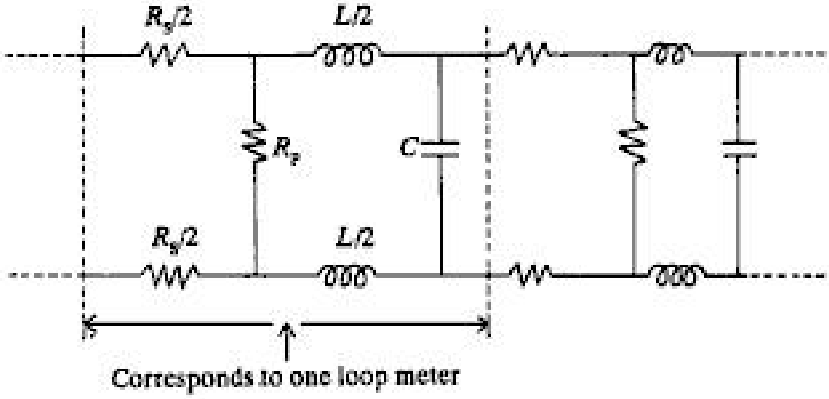

Fig. 5.18 Equivalent circuit of a transmission line.

resistance of the conductors, R_p the leakage resistance of the insulation, C the capacitance formed between the two conductors and L the inductance. R_s and L are values corresponding to one loop-meter and R_p and C are values corresponding to one running meter of the transmission line. Based on the equivalent circuit shown in Fig. 5.18, the general expression for V_p, A_c and Z_0 may be developed. But these are rather complex functions of R_s, R_p, C, L and frequency. However, for practical purposes, we may consider three

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number of bits b_n that can be transmitted reliably depends on the frequency tolerances of the clocks. The end of transmission is signalled by 1, $1\frac{1}{2}$ or 2 stop bits. The number of stop bits depends on the receive side equipment. The stop bits provide the minimum buffer period required by the receive equipment between two successive transmissions. Electromechanical devices require longer buffer periods than the electronic ones. In asynchronous transmission, which is also known as **start-stop transmission** for obvious reasons, the bit stream length is usually eight bits corresponding to a character. Hence, this transmission is often termed as character mode transmission. Two additional bits, one start and one stop bit for every eight bits represents a 25% transmission overhead. Therefore, it is not economical to transmit large volume data using asynchronous mode.

Synchronous transmission cuts down the transmission overhead by using a single clock for both transmission and reception as illustrated in Fig. 5.21.

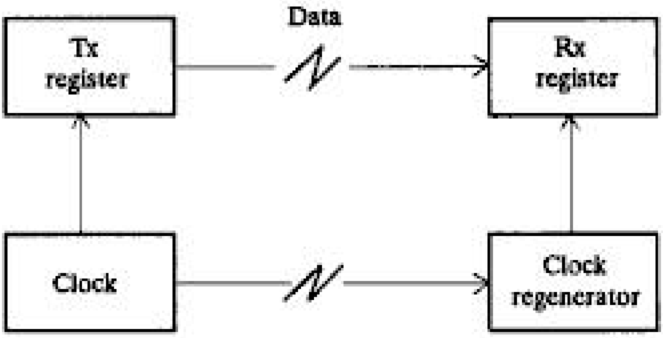
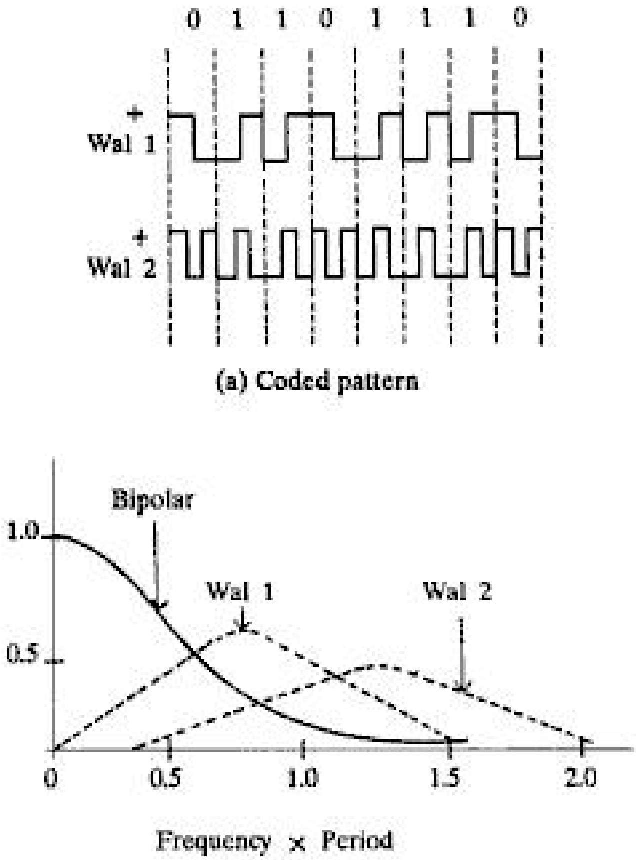


Fig. 5.21 Synchronous transmission.

Here, the transmitter clock is sent to the receiver which uses the same clock to sample the incoming data. Simple but an expensive way of sending the clocking information to the receiver is to use a separate channel for this purpose. Alternatively, synchronous receivers can be equipped with special tuned circuits that resonate at the desired clock frequency when excited by a received pulse or special phase locked loop (PLL) circuits that are capable of deriving the clocking information from the data itself.

The key to extracting clock from the data is in the signal transitions that occur in the data. If the signal is a continuous '1' or '0' for a sufficiently long time, the clock extraction process suffers and the receiver and transmitter may go out of synchronism. A number of techniques have been developed to ensure adequate density of signal transitions in the data transmitted in synchronous mode:

1. Restricting the source code
2. Use of dedicated timing bits



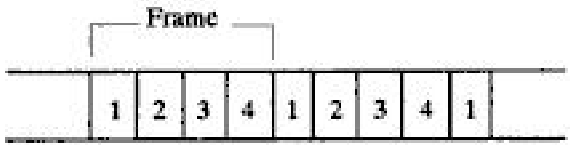
(a) Coded pattern

(b) Line spectra

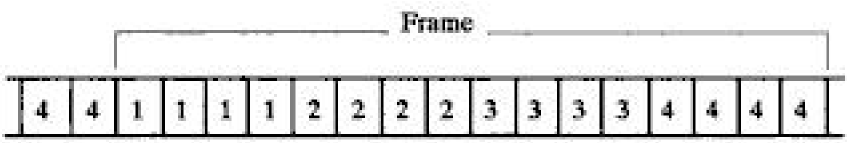
Fig. 5.24 Walsh codes.

components in the spectrum. The low frequency component is also insignificant as seen from Fig. 5.25(b). The code has some inherent error detection features. Errors result in AMI rule violation, which can be recognised at the receiving end. Deliberate violation of AMI rule can be made to send some control information from the transmitter to the receiver. AMI coding resolves two important problems encountered by many other bipolar codes. Firstly, there is no d.c. component in the transmitted signal and hence the phenomenon of d.c. wander is absent. Secondly, excellent timing information is available for continuous ones along with error detection capability. However, the code fails to provide timing information when a long series of zeros are transmitted.

In TDM, capacity allocation may be done either bitwise or wordwise. In bitwise allocation, each source is assigned a time slot corresponding to a single bit and in wordwise allocation, a time slot corresponds to some larger number of bits (often 8 bits) referred to as a word. The two allocation techniques give rise to two different TDM frame structures as shown in Fig. 5.27. In one case, the frame size is equal to the number of channels



(a) Bit interleaving



(b) 4-bit word interleaving

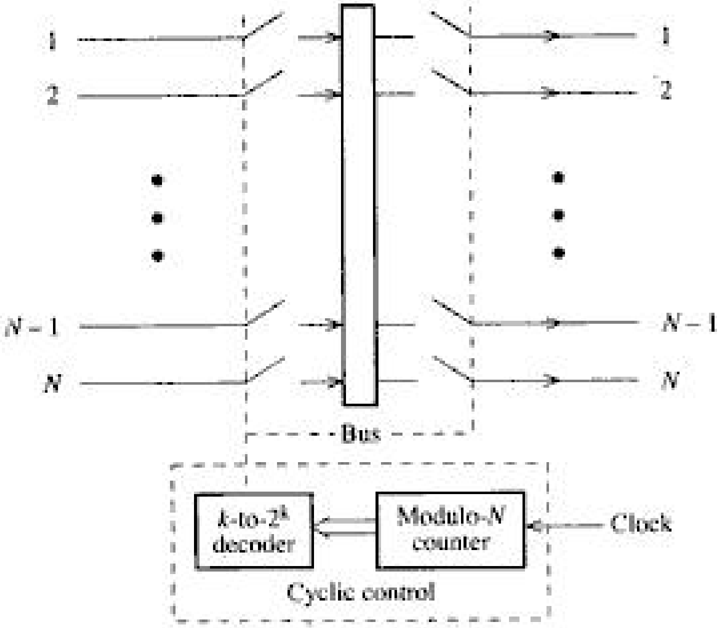
Fig. 5.27 Four-channel TDM frame structures.

multiplexed and in the other it is equal to the product of the word size and the number of channels. Bitwise interleaving is natural if delta modulator coders/decoders (codecs) are used and wordwise interleaving is natural if PCM codecs are used for digitisation. It may be recalled that in delta modulation, the coder produces a bit-by-bit output representing the slope variation whereas in PCM, sampled value is output as an 8-bit quantity. The bit rate of a TDM stream should be equal to or greater than the sum of the bit rates of the individual channels.

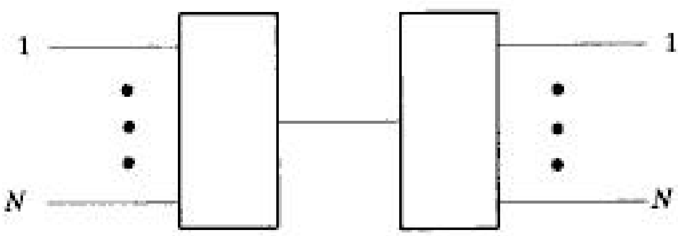
Having formed a frame by bit interleaving or word interleaving, it is necessary to have a mechanism to synchronise the frames at the transmitting and the receiving end, i.e. the start of the frame must unambiguously be identified at the receiving end. Frame synchronisation may be done in a number of ways. In all the ways, there are one or more framing bits with an identifiable data sequence. Frame bits may be added additionally to data bits or some of the data bits may be used as frame bits. For example, in Bell Systems T1 channel frame, an additional bit is introduced for every frame which alternates in value. The T1 channel structure multiplexes 24 channels and hence the frame has a length of 193 bits ($24 \times 8 + 1$). In one of the early

EXERCISES

1. The threshold of hearing or the reference level for the minimum discernible sound is internationally agreed upon as 10^{-12} W/m^2. The threshold of pain is 130 dB above this threshold of hearing. What is the actual power level at the threshold of pain?
2. The human ear has logarithmic response characteristics. How would a person perceive the change in loudness if he moves from a quiet residence which has an intensity level of 30 dB to an office with a level of 60 dB and an airport with a level of 120 dB?
3. List and discuss at least five advantages of digital transmission of speech over analog transmission.
4. What is the minimum sampling frequency required for a signal with a frequency range of d.c. –15 kHz if it is (a) band limited between 1 kHz and 10 kHz, (b) passed through a low pass filter which has a cut off frequency of 5 kHz.
5. The tone generated by push button 3 of a DTMF telephone is sampled at 1.6 kHz and passed through a PAM decoder which has a cut off frequency of 800 Hz. What frequencies will be present at the output of the PAM decoder?
6. How much does the SQR of a uniform PCM encoder improve when one bit is added to the code word?
7. If a minimum SQR of 33 dB is desired, how many bits per code word are required in a linearly quantised PCM system?
8. A 7-bit uniform PCM system has a bit rate of 56 kbps. Calculate the SQR when the input is a sine wave covering the full dynamic range of the system. Calculate the dynamic range of the sine wave input if the SQR is to be at least 30 dB.
9. If a linear PCM system is to have a minimum SQR as in the first segment of A-law companded PCM, calculate the number of bits required per sample. Assume a dynamic range equal to that obtained between the first and the eighth A-law segments.
10. What is the dynamic range provided by a uniform PCM encoder with 12-bit code words and a minimum SQR of 40 dB?
11. A PCM encoder using A-law has a dynamic range of ±2 V. Determine the code word when the input signal is (a) –120 mV, (b) 0.5 V, (c) –1.5 V, (d) 1.8 V.
12. For Problem 11, determine the input signal value if the code word is (a) 0110 1001, and (b) 1001 1011.



(a) Switching structure



(b) Two-stage equivalent.

Fig. 6.1 Simple PAM time division switching.

and $1 \times N$ switching matrices for the first and second stages respectively as shown in Fig. 6.1(b). The network has one link interconnecting the two stages. Each inlet/outlet is a single speech circuit corresponding to a subscriber line. The speech is carried as PAM analog samples or PCM digital samples, occurring at 125-μs intervals. When PAM samples are switched in a time division manner, the switching is known as **analog time division switching**. If PCM binary samples are switched, then the switching is known as **digital time division switching**. In Fig. 6.1(a), the interconnecting link is shown as a bus to which a chosen inlet-outlet pair can be connected by a suitable control mechanism and the speech sample transferred from the inlet

t_m = time to read the control memory

t_d = time to decode address and select the inlet or outlet as the case may be

t_t = time to transfer the sample value from inlet to outlet

All time values are expressed in microseconds. Equations (6.1) and (6.2) are valid for a 8-kHz sampling rate and a nonfolded network. The clock input to the modulo-N counter has a rate such that N sample transfers are organised in 125 microseconds. In other words,

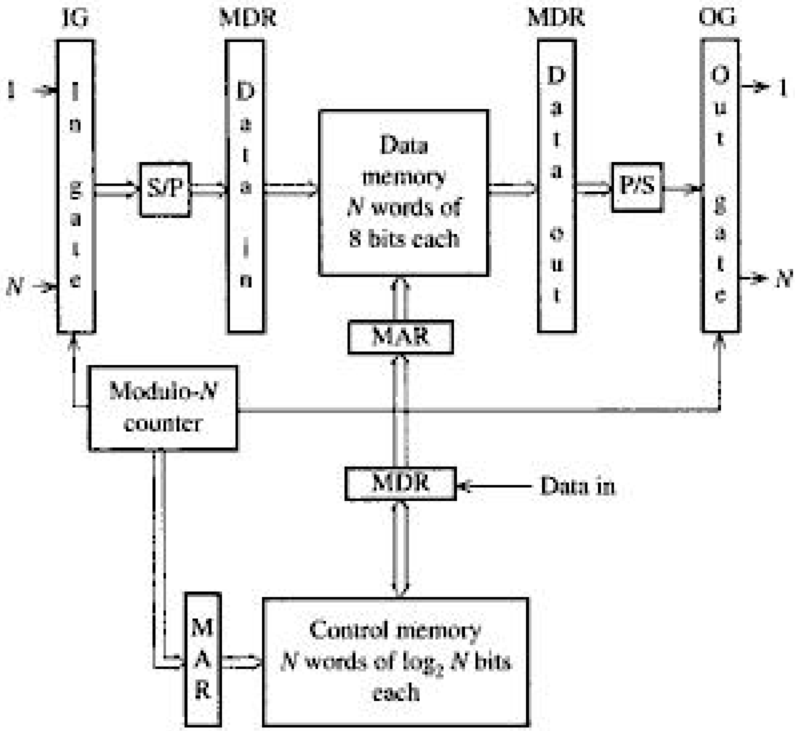
$$\text{Clock rate} = 8N \text{ kHz} \tag{6.3}$$

The output-controlled switches are capable of supporting broadcast connections, whereas the input-controlled ones are not. Broadcast takes place when all the control memory locations contain the same inlet address, in which case, the data from the specified inlet is transferred to all the outlets.

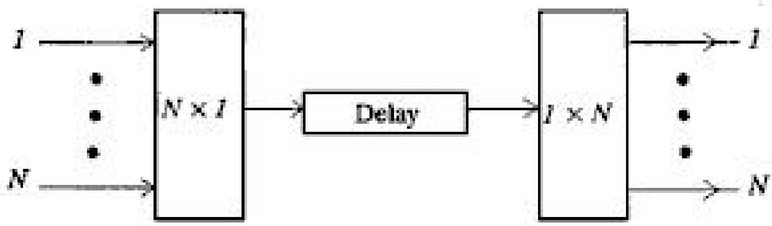
In the discussions so far, we have assumed that the speech samples are transferred from inlet to outlet. In practical telephone conversations, speech samples have to be exchanged both ways. For this purpose, two independent buses may be used on which data transfers take place simultaneously in opposite directions. Alternatively, a single bus may be used to organise the two-way data transfer first in one direction and then in the other. Digital buses, used with PCM samples, usually support parallel data transfer and if the incoming data is in serial form it is passed through a serial-to-parallel converter before being fed to the bus. Similarly, a parallel-to-serial converter is required on the output side.

The input or output-controlled configurations can be used to support folded network connections. Interestingly, these configurations, with a single bus, support two way transfers for folded networks. To illustrate this, let us consider a connection between subscriber 4 and subscriber 27 in an input-controlled switch. The control memory location 4 contains the value 27 and the location 27 contains the value 4. When the subscriber line 4 is scanned, by the cyclic control, the input sample from the *tip* (transmit) line of the subscriber 4 is transferred to the *ring* (receive) line of the subscriber 27. When the subscriber line 27 is scanned, the input from the line 27 is transferred to the *ring* line of the subscriber 4. Thus, data samples are exchanged both ways in one 125-μs cycle. When the switches are operated in this fashion, the Eqs. (6.1)–(6.3) apply to folded networks as well.

The fact that only $N/2$ simultaneous conversations are possible in a folded network can be used to arrive at another switch configuration with only $N/2$ control memory locations. In this case, no cyclic control is possible and both the input and output are memory controlled. But before discussing this configuration, we take a look at the design parameters of a time division space switch and compare the same with that of a space division switch. The definitions of the design parameters of a switch have been given in



(a) Switching structure.



(b) Equivalent circuit.

Fig. 6.5 Basic time division time switching.

1. Sequential write/random read
2. Random write/sequential read
3. Random input/random output.

In the first two methods of control, the sequential/random read/write operations refer to the read/write operations associated with the data memory. In

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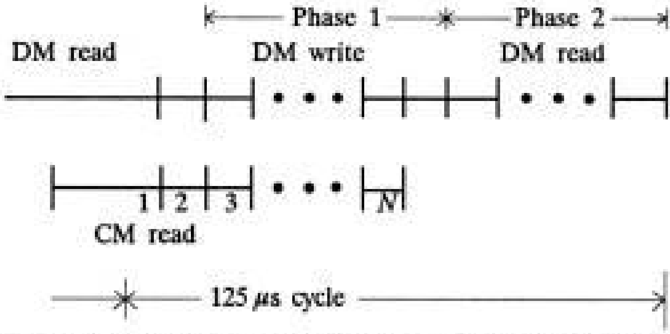

Fig. 6.7 Overlapping of data memory write and control memory read operations.

nonblocking. If we have a scheme whereby only the active subscribers are scanned, then the total number of subscribers connected to the system can be increased significantly.

Random input/random output form of control permits a larger number of subscribers than the switching capacity of the system. The switch in this case is, however, blocking in nature. The random input/random output scheme is illustrated in Fig. 6.8. Functionally, there are two control memory

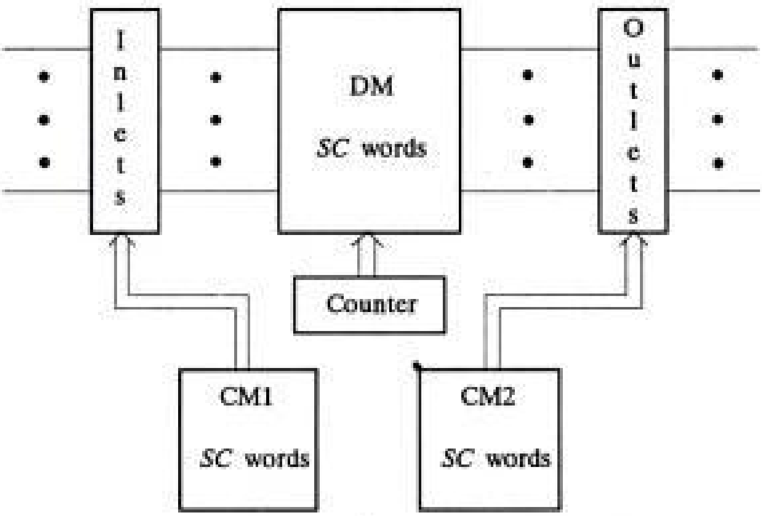

Fig. 6.8 Random input/random output time switch.

modules, CM1 and CM2, which hold the addresses of the active inlets and outlets respectively. There is a one-to-one correspondence between the locations of the two control memories. If the address of an active inlet is

taken out of the occupied list and vice versa. The actions involved in these operations are:

1. Allocate a location from the free list to the occupied list

Location to be allocated, F	=	contents of FLP
Contents of FLP	=	pointer in location F
Pointer in location F	=	contents of OLP location F
Contents of OLP	=	location F

Figure 6.9(b) shows the status of the lists after the first location in the free list is moved to the occupied list. The free list now is 9–3–10–5–4–1 and the occupied list is 7–2–11–12–8–6.

2. Free the location X from the occupied list and add to the free list

Pointer in the predecessor of X	=	pointer in X
Pointer in X	=	contents of FLP
Contents of FLP	=	X

Figure 6.9(c) depicts the list status after location 11 is freed from the occupied list and added to the free list. The new free list is 11–9–3–10–5–4–1 and the occupied list is 7–2–12–8–6.

6.3 Time Multiplexed Space Switching

In Sections 6.1 and 6.2, we dealt with time division switches where an inlet or an outlet corresponded to a single subscriber line with one speech sample appearing every 125 μs on the line. Such switches are used in local exchanges. We now consider switches that are required in transit exchanges. Here, the inlets and outlets are trunks which carry time division multiplexed data streams. We call such switches **time multiplexed switches**. In this section, we consider time multiplexed space switches and in Section 6.4 we discuss time multiplexed time switches.

A time multiplexed time division space switch is shown in Fig. 6.10. There are N incoming trunks and N outgoing trunks, each carrying a time division multiplexed stream of M samples per frame. Each frame is of 125-μs time duration. In one frame time, a total of MN speech samples have to be switched. One sample duration, 125/M microseconds, is usually referred to as a **time slot**. In one time slot, N samples are switched. Figure 6.10 shows an output-controlled switch. The output is cyclically scanned. There is a 1-to-M relationship between the outlets and the control memory locations, i.e. there are M locations in the control memory corresponding to each outlet.

The control memory has MN words. If we view the control memory as M blocks of N words each, a location address may be specified in a two dimensional form, (i, j), where i is the block address and j is the word within the block. We have $1 \le i \le M$ and $1 \le j \le N$. The block address i corres-

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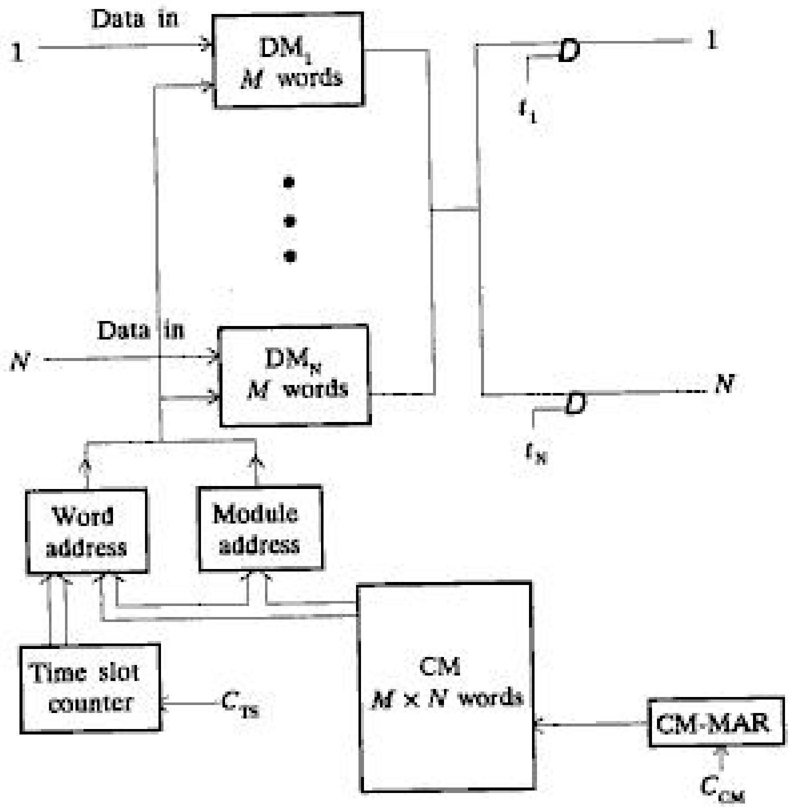
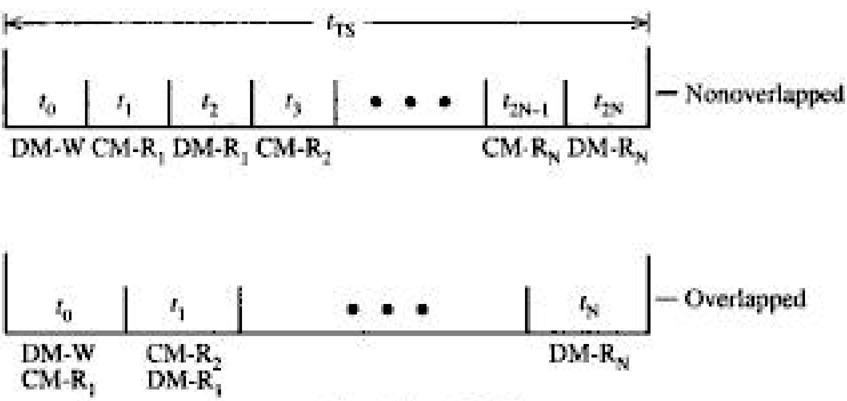


(a) Configuration



(b) Timing details

Fig. 6.16 Parallel-in/serial-out configuration.

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A time-space switch is shown in Fig. 6.18. The first stage consists of one time slot interchanger per inlet and the second stage a $N \times N$ space switch.

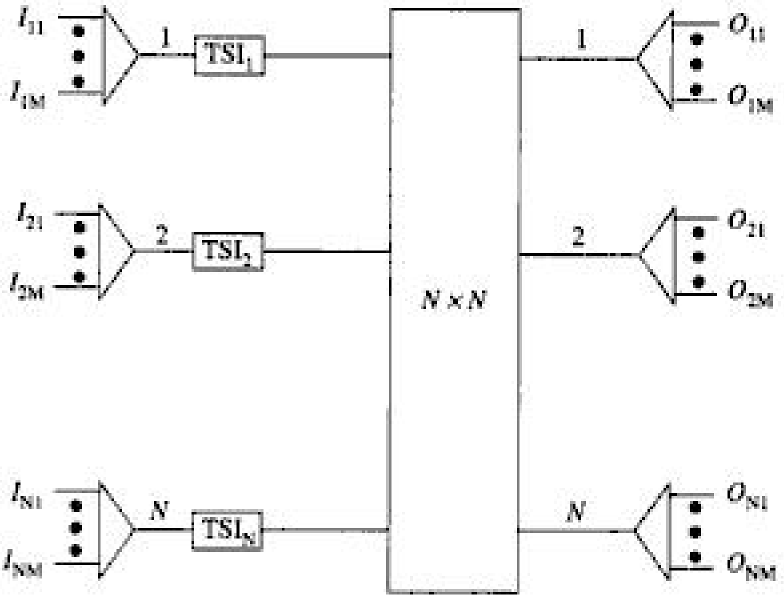

Fig. 6.18 Two-stage TS switch.

The control memories for the TSI and space switch are not shown in Fig. 6.18. Each time multiplexed inlet/outlet stream carries M channels. A subscriber on the input side is assigned to one of the inlets and a time slot in that inlet. An input subscriber assigned to line 4 at time slot 7 is identified by the label I_{47}. Similarly, a subscriber connected to the outlet 5 and time slot 6 is identified by O_{56}. The corresponding time slots are identified as IS_{47} and OS_{56}. Suppose that a communication is to be established between these two subscribers. The input sample from IS_{47} is first moved to IS_{46} at the output of TSI switch. During the time slot 6, a connection is established between the inlet 4 and the outlet 5 at the space switch. While this switch configuration ensures full availability, it is not nonblocking. Consider the two connections to be established between I_{47} and O_{29}, and I_{43} and O_{69}. Both the samples originate from the same inlet and are destined to the same time slot in different outlets. Both input samples require to be switched to time slot 9, which is not possible. Only one of them can be switched to slot 9. In general, blocking occurs if two inputs I_{ij} and I_{ik} are destined to outputs O_{pq} and O_{rq}. In other words, it is only possible to set up a single connection between any of the subscribers on an inlet and any of the subscribers connected to the same time slot on the output side.

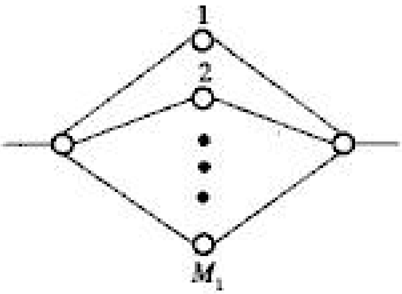

Fig. 6.21 Probability graph of a TST switch.

A space-time-space (STS) architecture consists of an $N \times k$ space matrix at the input, an array of k TSI switches in middle and a $k \times N$ space matrix at the output as shown in Fig. 6.22. In this architecture, the choice of input and

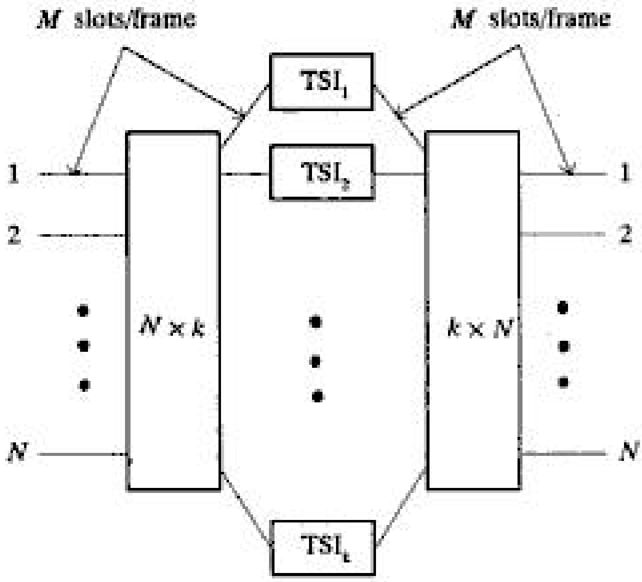

Fig. 6.22 A STS switch.

output time slots is fixed for a given connection. But the flexibility is provided by the ability to utilise any free TSI switch by space switching on the input and the output side. There are as many alternative paths for a given connection as there are TSI switches. For example, I_{79} may be connected to O_{84} by passing through TSI_3. The sequence is as follows:

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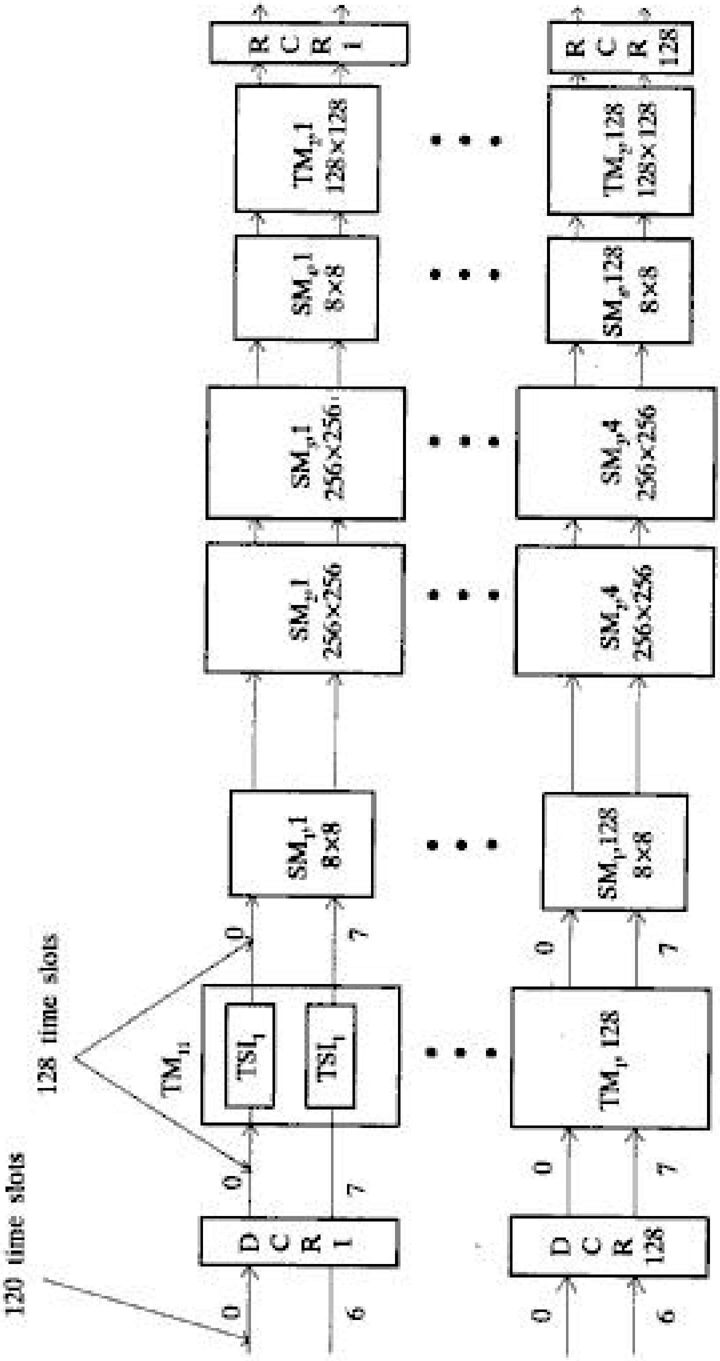

Fig. 6.23 No.4 ESS time and space stage.

15. A TSSTSST switch supports 1024 trunks each carrying 128 channels. Determine the basic clock rate of the switch if the input channels were to have 4096 alternative paths. The T stages are symmetrical. Is any useful purpose served by the middle T stage? Explain.

16. In n-stage combination switching a trade-off between blocking probability and time delay is possible. Explain.

17. Is TS nework nonblocking? Explain.

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The propagation of light along an optical fibre may be described in terms of a set of guided electromagnetic waves called the modes of the optical fibres. Each guided mode is a pattern of electric and magnetic field lines which is repeated along the fibre at intervals equal to the wavelength. Only a certain discrete number of modes are capable of propagating along the waveguide (fibre), depending upon the geometry of the fibre. The physical dimension and the configuration of the fibre may be chosen such that only one mode propagates through the fibre or many modes propagate. Accordingly, fibres are classified as **single mode fibres** or **multimode fibres**.

Mode types and refractive index types together give four different types of fibres, as shown in Table 7.1. Of these four, only three are practically

Table 7.1 Fibre Types

Fibre	Single mode	Multimode
Step index	√	√
Graded index	X	√

realisable. Restricting the propagation to a single mode demands that the core dimension is relatively small. When the core dimension is small, it is difficult to introduce graded refractive index. As a result, the graded index single mode fibres are not practicable as technology stands today.

The physical dimensions and the values of the refractive indices for commonly used telecommunication fibres are shown in Fig. 7.3. The core diameter used for single mode fibres lies in the range of 8–12 μm, and the diameter of the cladding lies in the range of 100–125 μm. A multimode fibre has a core diameter of 50–200 μm and a cladding diameter of 125–400 μm. The refractive indices of the core and cladding are related by the expression

$$n_2 = n_1(1 - \Delta) \tag{7.1}$$

where

n_2 = refractive index of the cladding

n_1 = refractive index of the core

Δ = a constant known as index difference

Beyond the cladding and the sheath, the refractive index is that of the air which has a value 1.0. From Eq. (7.1) the core-cladding index difference is given as

$$\Delta = \frac{n_1 - n_2}{n_1} \tag{7.2}$$

In the case of a graded index fibre, the most commonly used law for varying the refractive index from the centre to the surface of the core is a power law which is stated as

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fibre δ_t. For simplicity, we consider only plane waves that are polarised normal to the reflecting surface. Phase change due to reflections is given by

$$\delta_r = 2 \text{ arc tan } \frac{n^2(\cos^2\phi_1 - 1)^{\frac{1}{2}}}{n \sin \phi_1} \tag{7.9}$$

where $n = n_1/n_2$.

Phase change due to travel along the fibre is given by

$$\delta_t = n_1 ks \tag{7.9a}$$

where

k = propagation constant in free space
s = distance travelled by the wave along the ray

In order for a plane wave to propagate along the fibre, the phase of the twice reflected wave must be the same as that of the incident wave. In Fig. 7.6, the phase must be the same at points A and C. If this is the case, the wave superimposes constructively with itself. Otherwise, the wave interferes with itself and dies out. Therefore, the phase shift between the points A and C must be an integral multiple of 2π for proper propagation through the fibre. There are two refraction points, A and B, and a travel of two distances, AB and BC, between the points A and C. The distance $AB = BC = d/\sin\phi_1$. Therefore, from Eqs. (7.9) and (7.9a) the total phase shift is obtained as

$$2\pi M = 2\delta_r + \frac{2n_1 kd}{\sin\phi_1} \tag{7.10}$$

where M is an integer that determines the allowed ray angles for waveguiding.

In multimode fibres, light rays belonging to different modes take different paths for travel and hence take different times to reach the destination end of the fibre. A light ray that propagates straight down the axis of the fibre takes the least amount of time, and a light that strikes the core-cladding interface at the critical angle takes the longest time to reach the other end of the fibre. As a result, light rays representing a pulse of light energy reach the far end of the fibre at different times and cause spreading out of the pulse energy. This is called **modal dispersion** or **pulse spreading**. The difference between the travel times of the fastest and slowest light rays is called **pulse spreading constant** Δt and is usually expressed in nanoseconds per kilometre. Modal dispersion in optical fibres is analogous to phase distortion in copper cables discussed in Section 5.7, and can lead to intersymbol interference. Pulse spreading also results in amplitude reduction at the far end, giving rise to a power loss.

The refractive index of a material is wavelength dependent. If some light energy consisting of different wavelengths is launched into a fibre, the different wavelength components will travel through the fibre at different speeds,

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of the valence band. The recombination process must conserve both energy and momentum. The energy band diagrams shown in Fig. 7.7 are plotted against momentum. In Fig. 7.7(a), the minimum and the maximum energy levels of the conduction and valence bands respectively have the same momentum value. Hence, there is a direct recombination between electrons and holes. Semiconductor materials which exhibit energy level characteristics as shown in Fig. 7.7(a) are called **direct band gap semiconductors**. In Fig. 7.7(b), the maximum and the minimum energy levels occur at different momentum values. Here, a third particle must take part in the recombination process in order to conserve momentum since the momentum of a photon is very small. Lattice vibrations in the crystal known as **phonons** serve this purpose. Semiconductors that exhibit energy-momentum characteristics similar to the one shown in Fig. 7.7(b) are known as **indirect band gap materials**. Optical sources, both LEDs and ILDs, are constructed using direct band gap materials.

7.3.1 Light Emitting Diodes

In order to achieve a high radiance and a high quantum efficiency, it is necessary to confine the charge carriers (electrons and holes) and the optical emission to the active region of the *p-n* junction. Carrier confinement achieves a high level of radiative recombination. Optical confinement prevents absorption of the emitted radiation by the material surrounding the *p-n* junction. A variety of LED configurations have been investigated to achieve carrier and optical confinement. Of these, the most effective and widely used structure known as **double heterostructure** is shown in Fig. 7.8. A double heterojunction configuration may be fabricated to emit light through a surface or an edge of the structure. The surface-emit type shown in Fig. 7.8 is called **Burrus type LED**. The sandwich structure of Burrus LED is capable of confining both the carriers and the optical emission to the central active region. The differences in the energy band gaps and the refractive indices of adjoining layers confine the charge carriers and the optical field respectively to the recombination layer.

The injected carriers in the *p-n* junction form and recombine in pairs to satisfy the requirement of charge neutrality in the semiconductor crystal. As a result, the densities of excess electrons and holes are equal, i.e. $n = p$. When carrier injection stops, the carrier density returns to the equilibrium value. In general, the excess carrier density decays exponentially with time according to the relation

$$n = n_0 \exp(-t/\tau) \tag{7.12}$$

where

n_0 = initial injected excess electron density

τ = carrier lifetime

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The **optical bandwidth** is obtained by setting $P(\omega) = P_0/2$. That is,

$$P_0/2 = P_0[1 + (\omega_0\tau)^2]^{-\frac{1}{2}}$$

i.e. $1 + (\omega_0\tau)^2 = 4$, or

$$\omega_0 = \sqrt{3}/\tau \tag{7.18a}$$

It may be noted that the optical modulation bandwidth is apparently greater than the electrical bandwidth by a factor $\sqrt{3}$. In a practical circuit, the lower value determines the performance limit.

EXAMPLE 7.3 The minority carrier recombination lifetime for an LED is 5 ns. The d.c. optical output power is 320 W. Determine the 3 dB optical and electrical bandwidth and the optical power output at 40 MHz.

Solution From Eqs. (7.18) and (7.18a), we have

$$\omega_e = \frac{1}{5 \times 10^{-9}} = 200 \text{ MHz, and}$$

$$\omega_0 = \sqrt{3}\, \omega_e = 350 \text{ MHz.}$$

From Eq. (7.17a),

$$P\,(40\text{MHz}) = \frac{320 \times 10^{-6}}{[1 + (2\pi \times 40 \times 10^6 \times 5 \times 10^{-9})^2]^{\frac{1}{2}}}$$

$$= \frac{320 \times 10^{-6}}{(2.57)^{\frac{1}{2}}} = 200\ \mu\text{W}$$

The transient response of the LED is limited by the carrier lifetime, junction space charge capacitance and the diffusion capacitance. If the amplitude of the step current used to determine the transient response is large, the effects of capacitances become relatively small as compared to the effect due to carrier lifetime. In this case, the rise and fall times can be shown as:

$$t = \tau\,(\ln 9) \tag{7.19}$$

In an LED, the emitted power and the bandwidth of operation are inversely related. For a given injection level, the **power-bandwidth product** remains constant. An attempt to increase the bandwidth results in reduced power emission. Hence, the power-bandwidth product, *PBW*, is an important factor and can be shown to be:

$$PBW = \frac{J}{\tau}\,\frac{hf}{q}\,\eta \tag{7.20}$$

where

J = current density

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As the charge carriers flow through the semiconductor material, some electron-hole pairs recombine and get absorbed. On the average, the charge carriers travel a certain distance known as **diffusion length**, before they recombine. The diffusion lengths, L_n for electrons and L_p for holes, are related to the respective carrier lifetimes by the expressions

$$L_n = (D_n \tau_n)^{\frac{1}{2}}, \qquad L_p = (D_p \tau_p)^{\frac{1}{2}} \tag{7.29}$$

where D_n and D_p are constants known as electron and hole diffusion coefficients respectively.

EXAMPLE 7.4 In a silicon photodiode, estimate the carrier lifetimes of electrons, given that the diffusion length is 1 μm and the mobility of electrons is 0.15 m^2/Vs. Assume an ambient temperature of 27°C.

Solution The diffusion coefficients of carriers are related to the carrier mobilities by the Einstein relation as

$$D = \mu \frac{kT}{q}$$

where μ is the mobility of carrier, k is the Boltzmann constant and T is the temperature in degrees Kelvin. Therefore,

$$D_n = \frac{0.15 \times 1.38 \times 10^{-23} \times 300}{1.6 \times 10^{-19}} = 4.14 \times 10^{-3}\ \mathrm{m^2/s}$$

From Eq. (7.29),

$$\tau_n = \frac{L_n^2}{D_n} = \frac{10^{-12}}{4.14 \times 10^{-3}} = 0.24\ \mathrm{ns}$$

Optical radiation is absorbed in the semiconductor material according to the exponential power law

$$P(x) = P_0 e^{-\alpha x} \tag{7.30}$$

$$P'(x) = P_0 - P(x) = P_0 - (1 - e^{-\alpha})^x \tag{7.31}$$

where

α = absorption coefficient at a given wavelength

P_0 = incident optical power at the starting point

Equation (7.30) gives the radiation power density at a distance x and Eq. (7.31) provides the radiation power absorbed in the distance interval x. The absorption coefficient for a given semiconductor material is a function of wavelength and rises sharply for lower wavelengths. Equation (7.26) gives an upper cutoff value for the wavelength. At wavelengths much lower than λ_{th}, the material absorption may become significant and limit the useful

where

ε_s = permittivity of the semiconductor

A = cross-sectional area of the depletion layer

w = width of the depletion region

It may be noted that if the depletion layer is very thin, the junction capacitance is excessive.

EXAMPLE 7.6 In a *p-i-n* photodiode the depletion region has a width of 10 μm and a cross-sectional area of 10 μm^2. The diode is connected to a resistive load of 1 kΩ. If the load resistance and the junction capacitance are the dominating resistive and capacitive values respectively, determine the bandwidth of the diode circuit. Assume that the dielectric constant of the semiconductor is 1.1.

Solution

$$\varepsilon_s = \varepsilon_0 k$$

where

ε_0 = permittivity of free space = 8.85×10^{-12} F/m

k = dielectric constant of the semiconductor

Therefore,

$$\varepsilon_s = 9.74 \times 10^{-6} \text{ pF}/\mu\text{m}$$

$$C_j = \frac{9.74 \times 10^{-6} \times 10}{10 \times 10^{-6}} = 9.74 \text{ pF}$$

$$B = \frac{1}{2 \times 3.14 \times 10^3 \times 9.74 \times 10^{-12}} = 16.35 \text{ MHz}$$

7.4.2 Avalanche Photodiode

Signal-to-noise ratio is an important parameter in any receiver system. Noise is introduced into a receiver system from external sources as well as by the components of the receiver system. For example, the dark current of a photodiode contributes to the noise in the system. In an optical receiver system with a *p-i-n* photodetector, the electronic noise introduced by the load resistor and the amplifier circuits following the photodetector forms a dominant component of the total noise. It would, therefore, be beneficial to multiply the signal level within the photodetector itself. **Carrier multiplication** is a fundamental process in **avalanche photodiodes** (APDs). A *p-i-n* photodiode converts one photon to one electron and has a conversion efficiency of less than unity. In contrast, one photon of energy results in multiple carriers in the case of APD. Obviously, there is a gain G associated with an APD.

The carrier multiplication process is not itself free of noise. While on an average, each photon-generated carrier leads to M carriers at the end of the multiplication process, some carriers may produce more than M carriers and others less than M. This statistical nature of the process introduces further noise in the system. The result is that while the signal current is multiplied by the factor M, the r.m.s. noise level increases by a factor MF, where F is the noise factor. This noise factor is always greater than unity and an increasing function of M given by

$$F = M^x \tag{7.48}$$

where the index x lies between 0.2 and 1.0, depending upon the material and the type of carrier initiating the avalanche.

To conclude, APDs have a distinct advantage over *p-i-n* photodiodes for the detection of very low light levels due to their inherent carrier multiplication property. But they have several drawbacks too:

1. Their structure is more complex than that of *p-i-n* photodiodes.
2. The fabrication is difficult and hence they are more expensive.
3. Carrier multiplication is a random process and hence introduces additional noise into the system.
4. High reverse-bias voltages required are wavelength dependent.
5. Multiplication factor is temperature dependent. Hence temperature compensation circuits are required for stabilised operation.

7.5 Power Budget Analysis

System performance and cost constraints are important factors in any communication system. In Chapter 1, we briefly described the structure of an optical communication system. In the simplest form, a fibre optic communication system consists of a source with its associated modulator circuits, an optical fibre, and a detector with its associated receiver circuits. Performance of the system is evaluated by analysing the **link power budget** of the system and the cost is kept minimum by carefully selecting the system components from a variety of available choices. Various sources, detectors and optical fibres have already been discussed in the earlier sections of this chapter. In this section, we consider link power budget calculations.

In a fibre optic system, optical power loss occurs due to

1. source-to-fibre coupling losses,
2. connector loss,
3. splice loss, and
4. fibre attenuation.

The link power budget should provide for these losses and an additional power margin to allow for component aging and temperature fluctuations.

1. lateral misalignment of the axes of the fibre pieces,
2. gap or separation between the fibres,
3. angular misalignment, and
4. imperfect surface finishes.

These impairments are illustrated in Fig. 7.19. Loss due to lateral displacement is usually negligible if the fibre axes are aligned to within

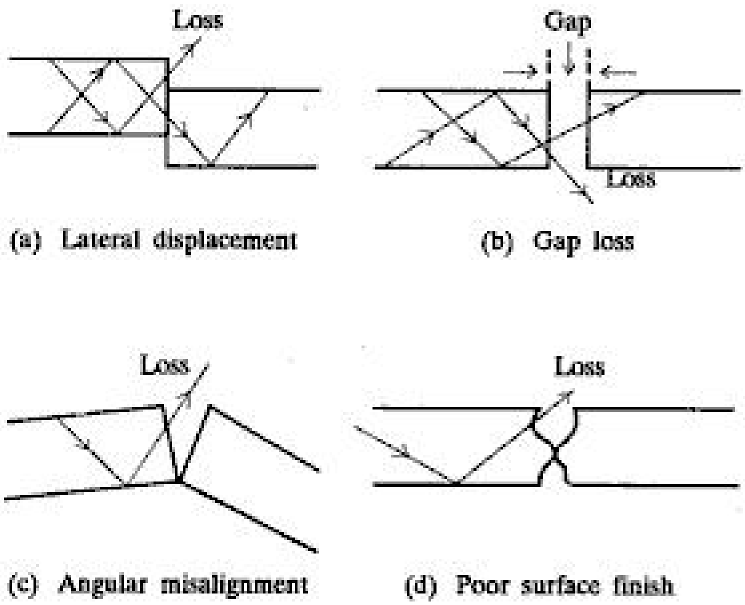

Fig. 7.19 Connector and splice losses.

five percent of the diameter of the smaller of the two fibres. Otherwise, the loss may run into several decibels. Splices are usually made by fusion of the two fibres. Hence, there are no gaps in splice joints. But, in the case of connector joints, the fibre ends are not made to touch each other to prevent damage to them due to rubbing while making and breaking a connection. The loss due to angular displacement is negligible if the misalignment is less than 2°. The ends of the two joining fibres should be highly polished and should fit together squarely. The imperfection tolerance in this case should be usually less than 3° off normal to ensure negligible losses.

Fibre attenuation is caused predominantly by the following factors:

1. Absorption loss
2. Scattering loss
3. Radiation loss.

Absorption loss is related to the material composition and the fabrication process of the fibre. The absorption loss may be caused by the

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the period during which the reduced tariff applies has been changed to begin later than 18.00 hours and one may expect the domestic call patterns also to change accordingly. During holidays and festival days the traffic pattern is different from that shown in Fig. 8.1. Generally, there is a peak of calls

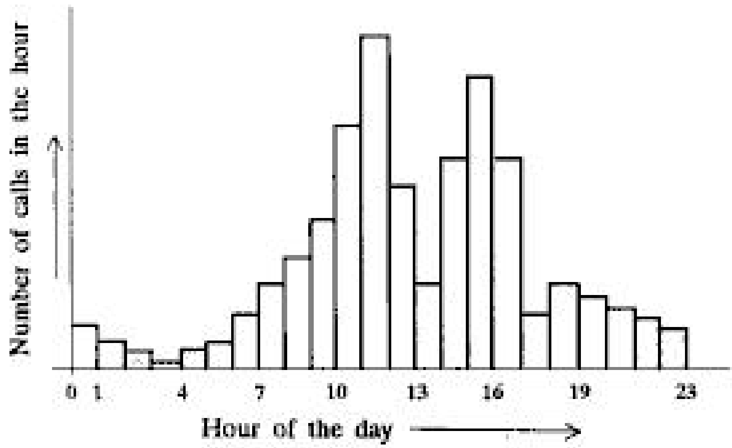

Fig. 8.1 Typical telephone traffic pattern on a working day.

around 10.00 hours just before people leave their homes on outings and another peak occurs again in the evening.

In a day, the 60-minute interval in which the traffic is the highest is called the **busy hour (BH)**. In Fig. 8.1, the 1-hour period between 11.00 and 12.00 hours is the busy hour. The busy hour may vary from exchange to exchange depending on the location and the community interest of the subscribers. The busy hour may also show seasonal, weekly and in some places even daily variations. In addition to these variations, there are also unpredictable peaks caused by stock market or money market activity, weather, natural disaster, international events, sporting events etc. To take into account such fluctuations while designing switching networks, three types of busy hours are defined by CCITT in its recommendations E.600:

1. **Busy Hour**: Continuous 1-hour period lying wholly in the time interval concerned, for which the traffic volume or the number of call attempts is greatest.
2. **Peak Busy Hour**: The busy hour each day; it usually varies from day to day, or over a number of days.
3. **Time Consistent Busy Hour**: The 1-hour period starting at the same time each day for which the average traffic volume or the number of call attempts is greatest over the days under consideration.

generated by the subscribers sometimes exceeds the network capacity. There are two ways in which this overload traffic may be handled: The overload traffic may be rejected without being serviced or held in a queue until the network facilities become available. In the first case, the calls are lost and in the second case the calls are delayed. Correspondingly, two types of systems, called **loss systems** and **delay systems** are encountered. Conventional automatic telephone exchanges behave like loss systems. Under overload traffic conditions a user call is blocked and is not serviced unless the user makes a retry. On the other hand, operator-oriented manual exchanges can be considered as delay systems. A good operator registers the user request and establishes connection as soon as network facilities become available without the user having to make another request. In data networks, circuit-switched networks behave as loss systems whereas store-and-forward (S&F) message or packet networks behave as delay systems. In the limit, delay systems behave as loss systems. For example, in a S&F network if the queue buffers become full, then further requests have to be rejected.

The basic performance parameters for a loss system are the grade of service and the blocking probability, and for a delay system, the service delays. Average delays, or probability of delay exceeding a certain limit, or variance of delays may be important under different circumstances. The traffic models used for studying loss systems are known as **blocking or congestion models** and the ones used for studying delay systems are called **queuing models**.

8.2 Grade of Service and Blocking Probability

In loss systems, the traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers. The overload traffic is rejected and hence is not carried by the network. The amount of traffic rejected by the network is an index of the quality of the service offered by the network. This is termed **grade of service (GOS)** and is defined as the ratio of **lost traffic** to **offered traffic**. Offered traffic is the product of the average number of calls generated by the users and the average holding time per call. This is given by Eq. (8.2). On the other hand, the actual traffic carried by the network is called the **carried traffic** and is the average occupancy of the servers in the network as given by Eq. (8.1). Accordingly, GOS is given by

$$\mathrm{GOS} = \frac{A - A_0}{A} \tag{8.3}$$

where

$$\begin{aligned} A &= \text{offered traffic} \\ A_0 &= \text{carried traffic} \\ A - A_0 &= \text{lost traffic} \end{aligned}$$

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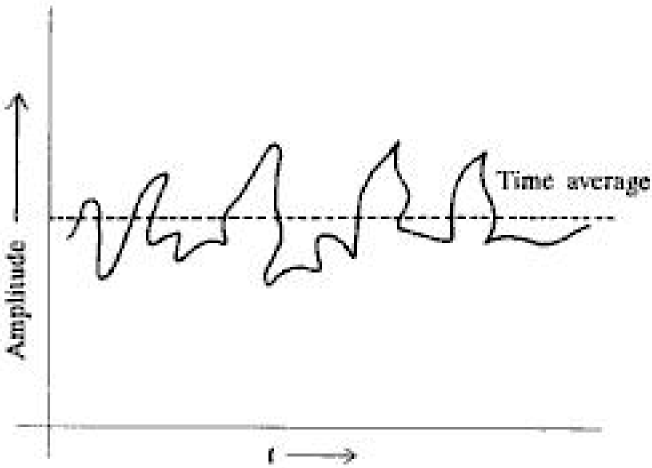
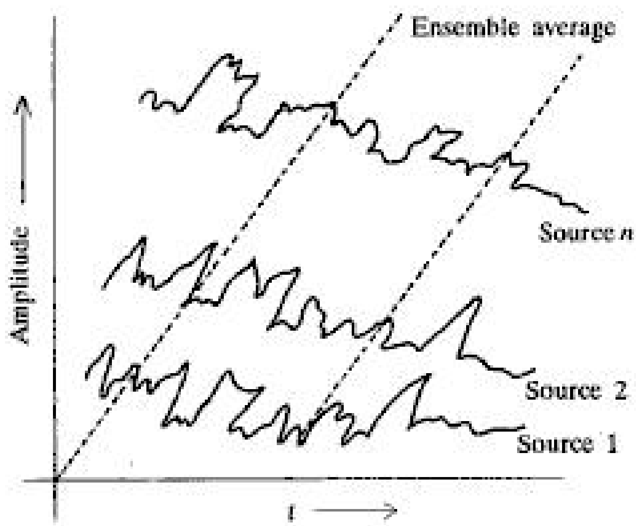
and the ones obtained using the second method are called **ensemble statistical parameters**. The word ensemble denotes the collection of sources. Figures 8.3(a) and 8.3(b) depict the time and ensemble averages of a stochastic process. These methods of obtaining statistical parameters of a



(a) Time average



(b) Ensemble average

Fig. 8.3 Time and ensemble averages of stochastic averages.

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In a system modelled as a B-D process, the termination phenomenon can be characterised as a **pure death process**. We obtain a pure death process from a B-D process by setting the birth rate equal to zero. We thus obtain the equations governing the dynamics of a pure death process from Eqs. (8.8) and (8.9) as

$$\frac{dP_k(t)}{dt} = \mu_{k+1} P_{k+1}(t) - \mu_k P_k(t) \quad \text{for } k \geq 1 \tag{8.17}$$

$$\frac{dP_0(t)}{dt} = \mu_1 P_1(t) \tag{8.18}$$

We can further simplify the behaviour of the pure death process assuming a constant death rate similar to the assumption we made with regard to the birth rate in the case of Poisson process. We then obtain the governing equations as

$$\frac{dP_k(t)}{dt} = \mu P_{k+1}(t) - \mu P_k(t) \quad \text{for } k \geq 1 \tag{8.19}$$

$$\frac{dP_0(t)}{dt} = \mu P_1(t) \qquad \text{for } k = 0 \tag{8.20}$$

The above equations can be solved by assuming suitable boundary conditions. We assume that at time $t = 0$ there are N members in the population. We then obtain the equations as

$$\frac{dP_k(t)}{dt} = \mu P_{k+1}(t) - \mu P_k(t) \quad \text{for } 0 < k < N \tag{8.21}$$

$$\frac{dP_N(t)}{dt} = -\mu P_N(t) \qquad \text{for } k = N \tag{8.22}$$

$$\frac{dP_0(t)}{dt} = \mu P_1(t) \qquad \text{for } k = 0 \tag{8.23}$$

Solving these equations, we get

$$P_N(t) = e^{-\mu t} \qquad \text{for } k = N \tag{8.24}$$

$$P_k(t) = \frac{(\mu t)^{N-k}}{(N-k)!} e^{-\mu t} \quad \text{for } 0 < k < N \tag{8.25}$$

$$P_0(t) = \frac{(\mu t)^{N-1}}{(N-1)!} e^{-\mu t} \quad \text{for } k = 0 \tag{8.26}$$

Equation (8.25) expresses the probability of no termination or death in a given interval as the population remains at the initial level. This is nothing but the probability distribution of how long the system remains in a state without a death or termination occurring. In other words, this is the probability

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blocking probabilities in finite source systems are always less than those for infinite source systems since the arrival rate decreases as the number of busy sources increases. Let

λ_s = arrival rate per subscriber
k = number of busy subscribers
N = total number of subscribers
R = number of servers

The offered traffic (arrival rate) when the system is in state k is given by

$$C_k = (N - k)\lambda_s \quad \text{for } k \leq 0 \leq R$$

The mean offered traffic rate is given by

$$C = \sum_{k=0}^{R} (N-k)\lambda_s P_k = N\lambda_s \sum_{k=0}^{R} P_k - \lambda_s \sum_{k=0}^{R} kP_k$$

$$= \lambda_s \left(N - \sum_{k=0}^{R} kP_k\right) \tag{8.40}$$

In Eq. (8.40), the quantity $\Sigma\, kP_k$ represents the average number of busy servers. The carried traffic in a network is the average number of calls accepted during the mean service time period. This is the same as the average number of busy servers at any given time. Hence, we have

$$C = \lambda_s (N - A_0) \tag{8.41}$$

The offered traffic is

$$A = Ct_h = \lambda_s t_h (N - A_0) \tag{8.42}$$

When the system is in state R, the offered traffic rate is $(N - R)\lambda_s$, but all the arrivals are rejected. Therefore, the lost traffic is obtained as

$$A - A_0 = (N - R)\lambda_s P_R t_h \tag{8.43}$$

The grade of service now works out to be

$$\text{GOS} = \frac{N - R}{N - A_0} P_R \tag{8.44}$$

Thus we see for Engest traffic, the blocking probability and the GOS are not the same, i.e. the time congestion and the call congestion values differ.

We now proceed to calculate the blocking probability. For this purpose, we analyse the steady state of the B-D process characterising this model. The arrival process has been discussed above. The termination process is the same as in the case of LCC model with infinite sources, and the termination rates is given by Eq. (8.33). Substituting the values of birth and death rates in Eqs. (8.10) and (8.11), we have

or

$$\frac{\text{Offered traffic}}{\text{Number of servers}} < 1$$

If this condition is not satisfied, the queue length would become infinite sooner or later, and the system would never be able to clear the traffic offered to it. A queuing system is characterised by a set of six parameters. A concise 6-parameter notation, due to D.G. Kendall, is used to represent different types of queuing systems. This notation uses letters to identify the parameters. The notation reads as $A/B/c/K/m/Z$. The parameter specifications are as follows:

A = arrival process specification
B = service time distribution
c = number of servers
K = queue capacity
m = number of sources (input population)
Z = service discipline

The value of the parameters K and m may be either a finite number or an infinite number. Queue discipline is the rule used for choosing the next customer to be serviced from the queue. Commonly used queue disciplines include first-come-first-served (FCFS), random selection, and priority based selection. The parameters K, m and Z may be omitted from the queue specifications, in which case they assume some default values. For K and m, the default values are infinity, i.e. infinite queue capacity and infinite sources respectively. The default queue discipline is FCFS. The parameter c is a nonzero positive finite number. The parameters A and B may assume any one of the values shown in Table 8.2. As an example, the queue specification $M/D/4$ means a queue system with Poisson arrival, deterministic service time distribution, four servers, infinite queue capacity, infinite sources and FCFS queue discipline. It is interesting to note that Kendall's notation may also be used to represent a loss system where the parameter K has a value zero.

When analysing delay systems in this section, we assume that infinite sources exist, infinite queuing capabilities exist, and the queue is serviced on FCFS basis. We assume a Poisson arrival process and concern ourselves with service times that are exponentially distributed or are constant. These service time distributions represent the most random and the most deterministic service times. Thus, a system that operates with some other distribution of service times has a performance somewhere between those produced by these two distributions. In effect, we shall model our telecommunication systems as $M/M/R$ and $M/D/R$ queuing systems. An important purpose of the analysis is to determine the probability distribution of waiting times and the associated mean values. Often, the probability that waiting time exceeds a specified limit is of interest. In particular, the probability that the waiting time is greater than zero represents the call congestion and hence is of immediate interest.

9. A traffic load of two erlangs is offered to a full availability group of five trunks. (Full availability implies that there is no restriction on the way in which calls are allocated to particular trunks). The average call holding time is three minutes.
   (a) What is the probability that no calls arrive during a five-minute period?
   (b) Determine the value of call congestion.
   (c) Consider the case when the trunks are numbered 1,2,3,4,5 and a call is allocated to the lowest numbered free trunk always. How much traffic is carried by the first trunk? How much traffic is carried by the last trunk?

   Assume LCC model.

10. Between two end offices there is an average traffic of 24 erlangs. If CCITT standard 32-channel PCM link is used between the end offices, what is the probability of blocking? How much of traffic is cleared by other resources if LCC model is assumed?

11. With Poisson arrival of two calls per minute, what is the probability that more than three calls will arrive in two minutes? What is the time during which at least four calls will arrive with a probability of more than 95 per cent?

12. A PABX with five trunks supports 200 extensions. Each extension generates three external calls per 8 hour working day. Average call duration is two minutes. If LCR model is assumed, what is the blocking probability? What is the offered load?

13. A PABX which is designed to be nonblocking internaly serves 100 extensions and has four trunk lines. Busy hour calling rate is two. Thirty per cent of the total traffic is external. Average call holding time is three minutes. What is the probability that an incoming call finds all the four trunk lines busy?

14. A PABX provides queuing and automatic call back facility for outgoing calls. If there are 20 outgoing call requests per hour and if the average call duration is three minutes, how many trunks are needed to ensure delays less than 30 minutes for 90 per cent of the requests?

15. A traffic arrival stream is formed by merging the input from K independent Poisson sources with source i having an arrival rate of λ_i for all $1 \leq i \leq K$. Show that the merged stream is also Poisson with an arrival rate of $\lambda = \sum_{i=1}^{K} \lambda_i$

16. An exchange uses two small call processors, each capable of serving requests that arrive at the rate of 15 requests per second. The two pro-

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Carrier systems employ multiplexing techniques and enable all the users to access the exchange over a single line. Analog FDM or digital TDM systems are used.

Signalling and voice transmission on the subscriber lines requires that the exchange performs a set of functions. These functions are performed by an interface at the exchange end known as **subscriber loop interface**. Some functions are required in analog networks, some in digital networks and others in both. The complete set of functions are known by an acronym **BORSCHT** which stands for:

B = battery feed
O = overvoltage protection
R = ringing
S = supervision
C = coding
H = hybrid
T = test

Functions B and R are well known. Overvoltage protection deals with equipment and personnel protection from lightning strikes and power line surges. Detection of off-hook condition is a supervisory function. Functions C and H are exclusive to digital switch interfaces. As we know, digital switching demands that analog-to-digital (A–D) and digital-to-analog (D–A) conversions and some form of coding/decoding be done. This aspect has been discussed in Chapter 5. Subscribers are connected to the exchange via 2-wire circuits. These circuits use a balanced connection as shown in Fig. 9.5. Balanced connections overcome many drawbacks of unbalanced

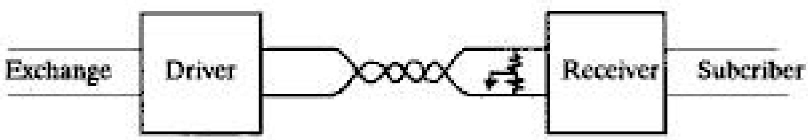

Fig. 9.5 Balanced circuit connection.

circuits. The transmission lines have equal impedances to ground and hence do not act as an antenna to pick up signals. Since the ground is not part of the signal path, hum is eliminated. Differential inputs improve noise immunity as any interference affects both lines equally and does not introduce differential currents.

Digital exchanges require receive and transmit signals on separate two-wire circuits. This calls for two-wire to four-wire conversion. Such a conversion is normally required for trunk transmissions in analog exchanges. The circuit that performs the 2-wire to 4-wire conversion is called **hybrid**. A transformer based hybrid circuit is shown in Fig. 9.6. The main function of a hybrid is to ensure that there is no coupling of signal from the input to the

P_4 = incoming power on the 4-wire circuit
P_2 = power reaching the 2-wire circuit
$P_4 - P_2$ = power reflected on to the return path

or, in terms of signal voltages,

$$RL = 20 \log \frac{V_4}{V_4 - V_2} \text{ dB} \tag{9.5}$$

$$= 20 \log \frac{1}{r_c} \tag{9.6}$$

where r_c is the reflection coefficient defined as

$$r_c = \frac{\text{reflected signal}}{\text{incident signal}} \tag{9.7}$$

If the two networks are perfectly balanced, then $Z_4 = Z_2$. Therefore, from Eq. (9.3) we have

$$RL \text{ (balanced)} = 20 \log \frac{2 Z_2}{0} = \infty$$

The return loss is infinite, i.e. the return signal experiences an infinite attenuation and hence there is no reflected signal.

EXAMPLE 9.3 In a national transmission system the characteristic impedances of the 4-wire circuit and the 2-wire circuit are 1000 Ω and 1200 Ω respectively. The average phase velocity of the signal in the circuit is 3×10^7 m/s. If the largest distance of a connection is 300 km, determine the attenuation to be inserted in the circuit.

Solution From equation (9.3) we have

$$RL = 20 \log \frac{2200}{200} = 20.8 \text{ dB}$$

$$\text{Round trip delay for echo} = \frac{300 \times 10^3}{3 \times 10^7} = 10 \text{ ms}$$

No echo suppressor is required in this case. From Fig. 9.9, for a round trip delay of 10 ms, we need about 7 dB loss to contain the echo. As per CCITT recommendations, the circuits would have been provided with a loss of 10 dB to control singing. Hence, no additional attenuator needs to be inserted in the circuit to control echos.

9.4 Transmission Systems

Modern long distance transmission systems can be placed under three broad categories:

isotropically radiated power (EIRP), where the gain of an isotropic antenna is taken as 1 or 0 dB. To distinguish the gain factor specified by Eq.(9.8) from power gain, this factor is called **directive gain**. It is important to note that power gain is more relevant in the case of transmitting antenna and directive gain in the case of receiving antenna.

An antenna can be mounted either vertically or horizontally depending upon practical convenience. Accordingly, a radiation pattern is said to be either **vertically polarised** or **horizontally polarised.** Since in skywave communication, the signal reaches the receiver by one or more reflections from the ionosphere, the angle of launch of signal into the ionosphere assumes importance. The path length is determined by this launch angle which is known as the **take-off angle** (TOA) and is defined as the angle between the vertical and the line of maximum radiation. For a given path length and horizontally polarised antenna, the TOA decreases as the height of the antenna increases and is given by

$$\text{TOA} = \sin^{-1}\left(\frac{\lambda}{4h}\right) \tag{9.11}$$

where λ is the wavelength of radiation and h is the height of the antenna from the ground.

Skywave communication is prone to **fading** which is of two types: general fading, in which the whole signal fades and selective fading in which only some of the frequency components of the signal fade. General fading can usually be handled by automatic gain control (AGC) mechanism of the receiver. Selective fading occurs when the skywaves reach the destination via two or more paths. Different path lengths lead to phase distortion at the receiving end. Use of single side band transmission or frequency modulation, or restriction of the propagation to only one mode are the methods by which selective fading effects can be minimised. Selective fading is also known as **multipath fading.**

EXAMPLE 9.4 An antenna has a directive gain of 12 dBi, a radiation efficiency of 90% and a feeder loss of 3 dB. Determine its power gain. Spell out the significance of the power gain and directive gain values.

Solution

$$\text{Radiation loss} = 10 \log (0.9) = -0.5 \text{ dB}$$

Therefore,

$$\text{Power gain} = 12 - 3 - (0.5) = 8.5 \text{ dBi}$$

The Power gain of this antenna gives an increase in field strength of 8.5 dB over isotropic antenna and its directive gain yields an improvement of 12 dB in received (S/N) ratio.

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By introducing processing and switching on board the spacecraft, dynamic time domain switching of traffic between multiple beams can be achieved, thus restoring full connectivity.

The cost of a satellite communication system is appropriated between the space and ground segments. The space segment comprises spacecraft and the equipment on board the spacecraft. The ground segment consists of the earth stations, antennas, ground control systems, user terminals etc. If the system is to serve a small population of ground terminals as in the case of intercontinental links, less investment can be made in the space segment and more in building large antennas and powerful ground stations. On the other hand, if the population of terminals to be served is large as in the case of direct broadcast systems, more investment is to be made in the space segment so that the cost of the terminals can be kept low. The present trend is to design more and more of **direct-to-user** (DTU) systems. This requires that we have satellite terminals that are affordable by every home. **Very small aperture terminals** (VSATs) are a step in this direction. Here, the nomenclature 'aperture' is used synonymously with the dish diameter. VSATs have dishes with apertures in the range of 1–2 m and are designed to be DTU terminals.

Small DTU terminals can, however, communicate only with large hubs and not directly amongst them. Direct VSAT-to-VSAT connectivity is not feasible. Figure 9.24 shows the communication scheme using a large hub and

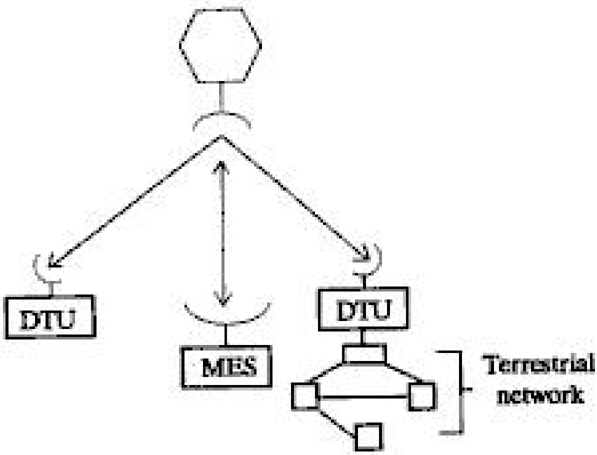

Fig. 9.24 Two-hop satellite network.

DTU terminals. There is a master earth station (MES) in the configuration which forms a vital communication link. The antenna of the MES acts as the large hub (typical size 10–30 m) with a high gain. A DTU terminal can communicate only with the MES antenna. Hence, information from one user to another is sent via the MES. The information hops twice before it reaches

single coaxial lines are called) are placed in a sheath. The characteristic impedance Z_0 of coaxial lines lies between 50 and 100 Ω. It is possible to estimate the line parameters of a coaxial line from its physical dimensions and the dielectric constant of the insulating material between the conductors. The line parameters of a transmission line have been discussed in Section 5.7, and we only present the values for a coaxial line here. Referring to Fig. 9.25, we have the following values for capacitance and inductance:

$$C = 24\varepsilon / \log(d_1/d_2) \ \text{pF/m}$$
$$L = 0.46 \log(d_1/d_2) \ \mu\text{H/m}$$

where ε is the dielectric constant of the insulating material. Therefore, the characteristic impedance Z_0 is given by

$$Z_0 = (L/C)^{\frac{1}{2}} = \frac{138}{\sqrt{\varepsilon}} \log\,(d_1/d_2)\ \Omega \tag{9.23b}$$

The attenuation constant A_c for a coaxial cable at an operating bandwidth of F MHz is given by

$$A_c = a + b + \sqrt{F} + cF \ \text{dB/km} \tag{9.24}$$

where a, b and c are constants dependent upon the physical parameters of the cable. For long haul transmission, two standard sizes are used. The dimensions of the standard cables and the associated values of a, b and c are presented in Table 9.7. The values of phase velocities V_p are also indicated for the two cables in Table 9.7 for high frequency (see Section 5.7) assuming PVC as the insulating material.

Table 9.7 Standard Coaxial Cables

	d_2 (mm)	d_1 (mm)	a	b	c	V_p (m/s)
Size 1	1.2	4.4	0.066	5.15	0.0047	1.8×10^8
Size 2	2.6	9.5	0.013	2.305	0.003	1.8×10^8

Coaxial cables are usually buried at a depth of 90–120 cm, depending on frost penetration in a given locality. Repeaters for coaxial cables are placed at uniform intervals along the route. To facilitate this, cable lengths are factory cut so that the splice occurs right at repeater locations. The repeaters are of two types: secondary or dependent repeaters and primary or main repeaters. Main repeaters have independent power sources and are installed in surface housing. Secondary repeaters derive their power from the cable itself and are usually buried. Power to the dependent repeaters is supplied by the main repeaters. A main repeater supplies power to about 15 secondary repeaters in each direction. The repeater spacing is dependent upon the size of the cable and the frequency of operation. For 2.6/9.5 mm cable at 12 GHz, the dependent repeater spacing is 4.5 km.

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Fig. 9.28 Indian telecommunication regions.

local, national and international calls. The term 'local call' here implies a call within a numbering area and the term 'national call' a trunk call between two different numbering areas within the same country. Basically, there are four possible approaches to dialling procedures:

1. Use a single uniform procedure for all calls, viz. local, national and international calls.
2. Use two different procedures, one for international calls and the other for local and national calls.
3. Use three different procedures, one for international calls, second for national trunk calls, and the third for local calls.

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limits its capability. The outband signalling suffers from the very limited bandwidth. Both are capable of having only a small signal repertoire. The trend in modern networks is to provide enhanced signalling facilities for the subscriber, the switching system and the telephone administration. Such a need is met by common channel signalling which may be implemented in two ways: **channel associated mode and channel nonassociated mode.** In the former, the common signalling channel closely tracks the trunk groups along the entire length of a connection. In the latter, there is no close or simple assignment of control channels to trunk groups. These modes are illustrated in Fig. 9.30. In the associated mode of operation shown in Fig. 9.30(a), the signalling paths for the speech paths *A-B*, *A-C-B* and *B-D* are *A-B*, *A-C-B* and *B-D* respectively. The term 'associated signalling' in the the CCS should not be confused with inchannel signalling. The signalling in CCS associated mode is still done on a separate signalling channel; only that the signalling path passes through the same set of switches as does the speech path. Network topologies of the signalling network and the speech network are the same. The advantages of the scheme are the economic implementation and simple assignment of trunk groups to signalling channels.

A CCS network consists of two types of nodes: **signalling transfer points (STPs) and signalling points (SPs)**. The signalling transfer points (STPs) usually have a connection with the switching centres although this is not essential. Since signalling originates from the control subsystems of the switching centres, these subsystems are referred to as SPs from the CCS signalling viewpoint. Signalling point is capable of handling control messages directly addressed to it but is incapable of routing messages. Signalling transfer point is capable of routing messages and could also perform the functions of a SP.

In the nonassociated mode of operation, the signalling information may follow a route that is different from the one taken by the speech path as shown in Fig. 9.30(b). The signalling paths for the speech paths *A-B* and *B-C* are *A-C-D-B* and *B-D-C* respectively. The network topologies for the signalling and the speech networks are different. There is no switching centre at the point *D* and only a STP is present. This approach offers flexibility as far as the CCS network is concerned. The signal messages may be transferred between the two end switching systems via any available path in the CCS network according to its own routing principles. This, however, demands a more comprehensive scheme for message addressing than is needed for channel associated signalling.

A CCS network may use associated signalling in some constituent parts and nonassociated signalling in other parts. The term 'quasiassociated signalling' is used to indicate such an operation. For example, if a signalling path exists between *B* and *C* in Fig. 9.30(b), the segment *B-C* may use associated signalling, whereas the signalling for the connection *A-B* is nonassociated. Quasiassociated signalling may also imply simplified signal

In SS5, the line signalling comprises either a compound of the two VF frequencies or a continuous single frequency. Interregister signalling uses 2-out-of-6 MF code. Initially, the system was jointly developed by UK Post Office and the Bell Laboratories for dialling over time assigned speech interpolation (TASI) equipped transAtlantic cables. This was the first application of intercontinental dialling and of TASI equipment. The system was subsequently specified by the CCITT as a standard in 1964 and has since found increasing applications in other parts of the world. Most of the Atlantic, Pacific and Indian Ocean circuits use SS5 at present. In view of the high cost of transocean cables and the consequent need to utilise the cables as efficiently as possible, TASI was considered essential for these cables. The design features of SS5 were dictated by TASI requirements.

The concept of TASI has been briefly explained in Section 8.5.4. We elaborate this now a little more. The normal speech activity of a subscriber on call is about 35 per cent. As a result, full duplex 4-wire speech transmission circuits are less than half utilised. The TASI technique attempts to improve trunk utilisation by assigning a circuit to a speech channel only when there is speech activity. In this way, a given number of circuits can support more than double the number of speech channels.

In TASI, each channel is equipped with a speech detector which, on detecting speech, arranges for a circuit to be assigned to that channel. Since this process of speech detection and establishment of trunk-channel association takes definite time, the speech burst is clipped for that duration. Typical clip duration is about 15 ms when a channel is available. It increases under busy traffic conditions when a free channel may not be available immediately. In order to reduce the extent of interpolation, a circuit is not disassociated from the channel for short gaps of speech. For this purpose, the speech detectors are provided with a short hangover time and a circuit is disconnected only when the speech gap is longer than the detector hangover time. The digital counterpart of TASI is known as **digital speech interpolation (DSI)**.

As with speech bursts, inchannel signalling information also experiences clipping in a TASI environment. This calls for special consideration in designing signalling systems for TASI environment. Unless signals are of sufficient duration to permit trunk-channel association and reliable recognition at the receiving end, there is the likelihood of the signal being lost partially or fully. With pulse signalling, it has been determined that a 500-ms duration is required to account for the extreme trunk-channel association condition. Allowing for reliable recognition, a pulse of 850 ± 200 ms duration is considered suitable. But, pulse signals of such lengths would slow down the signalling process considerably. The pulse gaps would result in the channel being disassociated, thus leading to unnecessary TASI activity. Moreover, fixed length pulses cannot take advantage of lightly loaded conditions when the channel assignment time is low. For these

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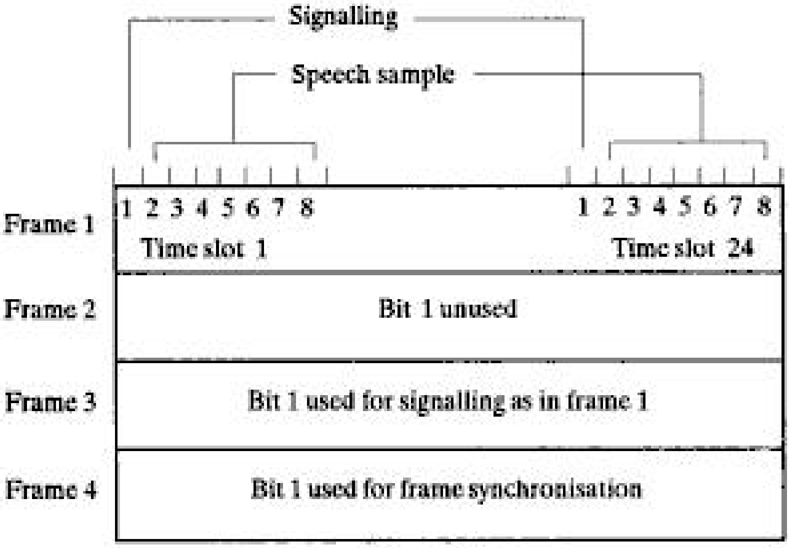

Fig. 9.33 Bell D2 24-channel multiframe PCM signalling structure.

alignment bits, i.e. 193rd bit in each frame, in the odd numbered frames are used for frame synchronisation and the ones in the even numbered frames for multiframe synchronisation. The signalling rate is 650 signals per second with two bits per signal.

The bit stealing in frames 6 and 12 in the Bell D2 system denies the full gain of 8-bit speech encoding. The outslot 30-channel PCM signalling system realises the full potential of 8-bit encoding as all the speech samples are encoded using eight bits. Of the 32 time slots per frame in this system, time slot 0 is used for frame synchronisation and the time slot 16 for signalling. Obviously, one time slot with eight bits cannot support signalling for 30 speech channels in every frame. Hence, a multiframe structure with 16 frames is adopted for signalling purposes. With this structure, time slot 16 in each frame except frame 0, carries signalling information for two speech channels. Thus, 15 time slots in frames 1–15 carry signalling information for 30 channels in every multiframe. With 15 time slots for 30 channels, four bits are available for signalling information per channel. The signalling rate works out to be 500 signals per second with each signal represented by four bits. The first four bits of time slot 16 in frame 0 are used for multiframe alignment with the bit pattern 0000, with the remaining four bits unused. To enable easy recognition, this all-zero pattern is prohibited as a combination of signalling bits in time slot 16 of other frames. This limits the number of signalling patterns to 15 per channel. Sometimes, this CEPT signalling scheme is referred to as **bunched signalling**.

microprocessor at the base station informs the master processor of this, which checks with the adjoining cell stations to find out which one of them is receiving a good (*S/N*) ratio at the present carrier frequency of the mobile station. The call is then transferred to this station in the adjoining cell. The transmit and receive frequencies of the user terminal are automatically changed to the values assigned by the new base station. The telephone exchanges directly associated with the call switch circuits to the new base station under instructions from the master station. Although the process of 'handing over' a mobile terminal from one cell to another is an involved one, the entire process of switching is completed within a few milliseconds such that a subscriber hardly notices any interruption in the call. There can, however, be difficulties if data is sent over cellular links as even a brief break in the link can affect data transfer.

Cellular communication service is more expensive than regular telephone service. The cellular mobile units are more complex in design than the regular telephone instruments as they need to have automatically tunable transmitters and receivers covering the entire frequency bandwidth. The design of the base stations and the master station is also complex.

A car cellular unit has three pieces of equipment: the transmission unit, the control unit and the antenna. The transmission unit is usually placed in the trunk of the car, the control unit is mounted on the dash board and the antenna is either roof-mounted or wind-shield mounted. The control unit resembles an ordinary telephone in design. Researchers are still looking for technological advancements to reduce the risk attempting to dial while driving a vehicle. Voice-activated dialling appears to be promising.

The development currently underway in the field of cordless phones is likely to further promote the cellular communications. The second generation cordless phones, called CT2, use digital technologies and are capable of working with very many CT2 stations which are to be established in railway stations and other public places. It is expected that by mid 1990s many CT2 stations will be in operation, variously called **phone zone, phone point, telepoint** etc. In working, CT2 phones send out their own identification code to the nearest phone point before each call to enable the exchange to bill the owner for the call. In order to reduce the complexity associated with the cellular systems, two restrictions are placed on CT2 systems. Firstly, no incoming calls are permitted to CT2 phones, so that the network does not have to keep track where a particular subscriber is at any point of time. Secondly, once a call is established, a subscriber is not permitted to move from one phone zone to another so that there is no need for the intercell handing over arrangements which complicate the cellular communications. Although research and development in cellular systems have been driven by the urgent need to improve mobile communications, the use of digital radio links on a very large scale with ordinary static telephone systems, replacing local distribution cable networks, may well turn out to be the most significant outcome of this research in the near future.

Intercity, intercountry and intercontinental networks are known as WANs. Based on the communication infrastructure used, they may be classified as **terrestrial data networks (TDNs)** or **satellite based data networks (SBDNs)**. In TDNs, data communication is organised using cables, fibre optic lines or radio links. A geostationary or a geosynchronous satellite is used for communication in SBDNs.

A metropolitan area network interconnects computers within a metropolitan city. Community antenna television (CATV) cables, twisted pair wires or shielded lines, optical fibres, radio links or line-of-sight (LOS) optical communication links provide the communication medium for MAN. The broadband capability of CATV cables permits carrying voice, data and video simultaneously. Thus MANs are usually multimedia networks.

Local area networks are confined to a single building or a group of buildings generally belonging to the same organisation. Optical fibres, twisted pair and coaxial cables are used as the communication media for LANs. Fibre optic networks (FONs) are suitable for both LANs and MANs. Synchronous optical networks (SONET) are designed to operate at high speeds.

LANs, MANs and WANs are generally interconnected in a hierarchical manner to form a global network as shown in Fig. 10.1. LANs are often

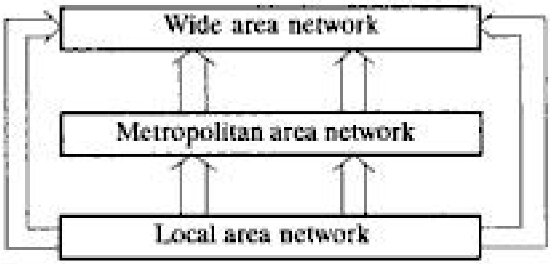

Fig. 10.1 Data network hierarchy.

connected directly to WANs, particularly in places where MANs are not installed or have not developed well. Apart from the different geographical coverages, the range of data rates supported on these networks also differs widely. Figure 10.2 summarises the typical data rates and geographical coverages for these networks.

10.1 Data Transmission in PSTNs

Public switched telephone networks and electronic PABXs are designed to carry analog voice signals. They can, however, be used for data transmission by employing suitable interfaces. As seen from Fig. 10.2, LANs can be designed around PABXs, and MANs around PSTNs. In these cases, the data rates are usually limited to a maximum of 64 kbps. Terrestrial data networks

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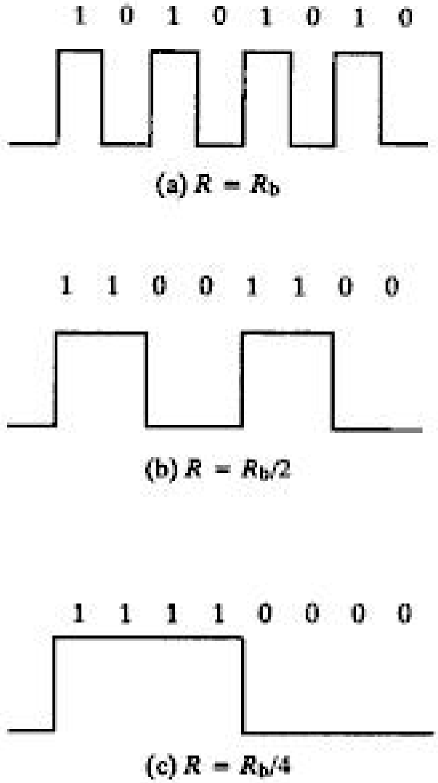

Fig. 10.4 Baud rates and bit rates.

using additional signal levels, say 16, the effective bit rates may go up to 9.6 kbps. The techniques of using additional signal levels to increase the bit rate are the ones used in the design of modems.

10.1.2 Modems

Amplitude, frequency and phase modulation are all used in the design of modems. In amplitude modulation, zeros and ones are represented by two different voltage levels. A signal waveform $s(t)$, called baseband signal, is generated from the digital data. This is then multiplied by a sinusoidal carrier, say $\cos(2\pi f_0 t)$, to generate a modulated signal

$$m(t) = s(t)\cos(2\pi f_0 t) \tag{10.3}$$

At the receiver end, the modulated signal is again multiplied by $\cos(2\pi f_0 t)$, yielding a received signal

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In addition, AFI also identifies the format and type of address present in the initial domain identifier (IDI) field and the domain specific part (DSP). Binary and packed decimal formats are generally supported. IDI and DSP types may include PSTN numbering, packet switched network numbering, ISDN numbering, telex numbering and other similar numbering schemes. The full SAP address is of variable length, up to 40 decimal digits or 20 bytes. An important objective of SAP addressing format is to enable convenient internetworking.

OSI reference model proposes a general layered concept, with provision for adding or deleting layers as demanded by factors like service complexity, technology options etc. Taking into account the functions required of the architecture for organising computer communication in the present day context, Sub Committee 16 of the Technical Committee 97 (TC97/SC16) of ISO, has recommended a 7-layer architecture, which is shown in Fig. 10.17. In the figure, two end systems that communicate with each other via two intermediate systems are shown. Only the first three layer functions come into action in an intermediate node. Entities in these layers always communicate with peer entities in the adjacent system. In other words, in the first three layers, the communication proceeds on a **link-by-link basis**. In contrast, entities in layers 4–7 communicate with peer entities in the end systems. There is no communication with entities in the intermediate systems. In this sense, layers 4–7 are often called **end-to-end layers**. The 7-layer architecture has been arrived at after a careful application of a broad set of layering principles. The important principles are:

1. Create layers to handle functions which are manifestly different in the process performed or technology involved.
2. Collect similar functions into the same layer and create a boundary at a point where the number of interactions across the boundary are minimised.
3. Create a layer of easily localised functions so that the layer could be totally redesigned and its protocols changed in a major way to take advantage of new advances in architectures, hardware and software technology without changing the services offered or the interfaces with the adjacent layers.

Sections 11.4 and 11.5 describe the functions performed, services offered, and the protocols used by each of the seven layers.

10.4 Link-to-link Layers

The first three layers, viz. physical, data link and network layers, form the link-to-link layers of OSI reference model. Entities in an OSI layer perform certain functions to fulfill the stated purpose of the layer. They obtain services from the immediate lower layer and provide services to the

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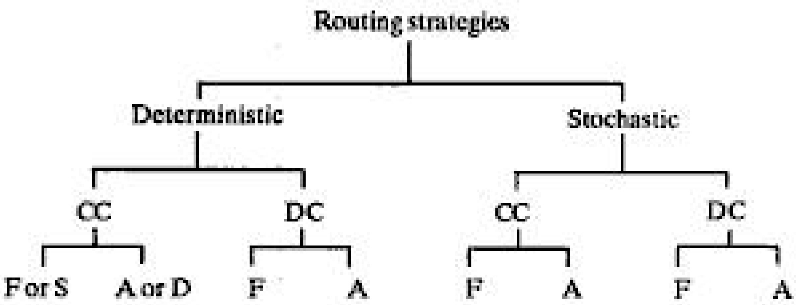
any routing or congestion control algorithms and they are left as implementation dependent features. We briefly discuss some of the generic routing and congestion control strategies here. Routing strategies may be classified as shown in Fig. 10.24. Routing algorithms may be deterministic or



CC = centralised control DC = distributed control F = fixed
S = static A = adaptive D = dynamic

Fig. 10.24 Classification of routing algorithms.

stochastic in nature. Routing decisions may be taken centrally by collecting global information from the network or in a distributed fashion based on the local conditions. Finally, each strategy may use a fixed criterion or adapt itself to varying network traffic conditions. A routing algorithm that uses a precomputed route from a given node to another is a deterministic, distributed, static algorithm. If the route changes, say with time of the day, one may call the algorithm dynamic but still deterministic. If the routing information is computed by a central node based on the knowledge of network topology etc. and distributed to other nodes, then it is centrally controlled. If the route is chosen based on some probability calculation, say a random number, the algorithm is stochastic in nature.

A number of measures may be used in assessing the performance of a routing algorithm:

1. Minimum delay
2. Minimum number of intermediate nodes or hops
3. Processing complexity
4. Signalling capacity required on the network
5. The rate of adaption in the case of adaptive algorithms
6. Fairness to all types of traffic
7. A reasonable response time over a range of traffic intensities
8. Robustness: the ability to reach the destination even when parts of the network fail
9. Stability: the ability to reach the destination quickly without wandering.

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4. A LAN adhering to a certain standard permits multivendor systems to be connected to it. Thus, a user is not committed to a single vendor.
5. In LAN, the systems are generally so chosen as to meet most of the user requirements locally and the network is used only for resource and information sharing purposes. Due to this, each user gets a better response than would be the case in a centralised system. LAN tends to exhibit an improved performance.
6. LAN offers flexibility in locating the equipments. Most computers on a LAN are physically placed at the user table, which is most convenient for working and improves the productivity significantly.

LANs also suffer from certain disadvantages:

1. The incremental growth may lead to uncontrolled growth and may ultimately result in more investment than would have been the case with a centralised system.
2. LAN standards offer many options. Although systems from different vendors conform to the same standard, options exercised may vary from vendor to vendor with the result the interoperability is not guaranteed. Incompatibility may arise at the hardware, software or data organisation level.
3. A distributed system environment raises problems of security, privacy and data integrity. For example, a computer virus introduced in one system may very quickly spread to other systems.

10.7.1 LAN Technologies

There are three major aspects in a LAN:

1. Medium of transmission
2. Topology
3. Access method.

Three important media of transmission are in use: twisted pair wire, coaxial or CATV cable, and fibre optic cable.

Twisted pairs are used in low speed LANs using baseband transmission. In this mode of transmission, data is transmitted as simple electrical levels often without any modulation. There is no multiplexing and the entire bandwidth of the medium is used for transmitting signals from one station. Baseband transmission links are sometimes termed 'wire only' links as there are no other equipments used for transmission.

In contrast, broadband transmission uses modulation techniques and is suitable for transmitting high speed and multiplexed data. CATV and coaxial cables are used for broadband transmission at speeds of 10 Mbps or more. Fibre optic cables carry data at rates up to 100 Mbps. Radio communication may also be used for transmitting data in LANs. Although some laboratory systems have been built using radio links, such systems are not in use.

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EXERCISES

1. There is a saying that "If Mohammad does not go to the mountain, the mountain comes to him". Discuss the analogy of this statement to remote computing.
2. Depict by means of a tree structured diagram the different classes of data networks.
3. A noiseless communication channel has a bandwidth of 6 MHz. What is the maximum data rate that can be supported in this channel if four levels are used to represent digital data?
4. What is the maximum data rate achievable, if a binary signal is sent over a 3-kHz channel whose S/N ratio is 40 dB?
5. If the CCITT standard 2048 kbps PCM stream is to be carried on a 50-kHz channel, what is the maximum tolerable S/N level?
6. The operating state diagram of a modem has two operating points on the x-axis. What modulation is used by the modem?
7. The operating state diagrams of a modem has states at the coordinates (1,1), (1,–1), (–1,–1) and (–1,1). What is the data rate achieved by the modem at 1200 baud?
8. A DCE converts an asynchronous transfer from the computer to synchronous transmission on the line. Asynchronous transmission uses one stop bit. If 100 characters are packed into a synchronous block, determine the transmission efficiency achieved by the DCE. Assume four bytes of overhead for every block.
9. List at least six differences between the postal and telephone systems and bring out the analogy between S&F and circuit switched connections.
10. A packet switching network has N nodes fully connected. What are the best, the average and the worst case transmission path lengths in hops?
11. A network offers both circuit and packet switching facilities. Given that:

    T_s = circuit set up time per station

    T_q = processing and queuing delay in each node for packets

11

Integrated Services Digital Network

Integrated Services Digital Network (ISDN) has been perhaps the most important development to emerge in the field of Computer Communications in the 1980s, and it will probably continue to dominate the developments in the 1990s too. Unlike many other developments, ISDN is a well conceived and planned area of development in the field of telecommunications. CCITT has been pioneering and guiding the efforts towards the development of ISDN. The process of digitalisation of telecommunication networks started in early 1960s when PCM was introduced for digital transmission of voice signals. Arrival of transistorised minicomputers opened up the possibility of digital switching in the late 1960s. CCITT was quick to recognise the feasibility of digital telecommunication networks and set up a study group called **Special Study Group D** in 1968 to look at a variety of issues related to the use of digital technology in telephone networks. This study group is the forerunner of today's Study Group XVIII set up in 1976, and has the responsibility for all ISDN related activities within the CCITT. The first formal definition of ISDN was given by the Study Group D in its recommendations, G.702, issued and adopted by CCITT in 1972:

> **Integrated Services Digital Network** An integrated digital network in which the same digital switches and digital paths are used to establish different services, for example, telephony and data.

It is interesting to note that a study, which started with the objective of exploring the use of digital technology, quickly shifted its emphasis to an Integrated Digital Network (IDN) and within a period of four years even foresaw the emergence of ISDN. From then on, the progress towards ISDN has been steady and unrelenting. In 1980, the first set of ISDN standards, G. 705, emerged, which laid down six conceptual principles on which ISDN should be based:

1. ISDN will be based on and will evolve from the telephony IDN by progressively incorporating additional functions and network features including those of any other dedicated networks so as to provide for existing and new services.

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ISDN architecture and a plethora of services that can be supported in ISDN. However, the ISDN technology, design approaches and standards are still evolving and will remain the focus of development in the 1990s.

Thus, we are now at a point of time when there exists a synergy of social demands, technological feasibility and economic viability with regard to ISDN. Consequently, an era of ISDN will be heralded all over the world during the next twenty years or so. As must be obvious by now, ISDN is the apt nomenclature for the telecommunications network of the future in which all network functions will be handled in digital domain and a single network will support a set of integrated services.

11.2 New Services

ISDN will support a variety of services including the existing voice and data services and a host of new services. A short list of some of the important new services is:

1. Videotex
2. Electronic mail
3. Digital facsimile
4. Teletex
5. Database access
6. Electronic fund transfer
7. Image and graphics exchange
8. Document storage and transfer
9. Automatic alarm services, e.g. smoke, fire, police and medical
10. Audio and video conferencing.

A few of the services are described in the following sections.

11.2.1 Videotex

Videotex is a generic term for systems that provide easy to use, low cost computer based services via communication facilities. Three forms of videotex exist:

1. View data
2. Teletext
3. Open channel teletext.

View data is fully interactive videotex. This means that requests for information or service from a user are actually sent to, received by, and acted on by a centralised computer.

Teletext is broadcast or pseudo-interactive videotex service. Teletext users may select the information to be seen, the pace at which the information is to be displayed and, often, the sequence of display. The information is cast in the form of frames and a set of frames which is called a

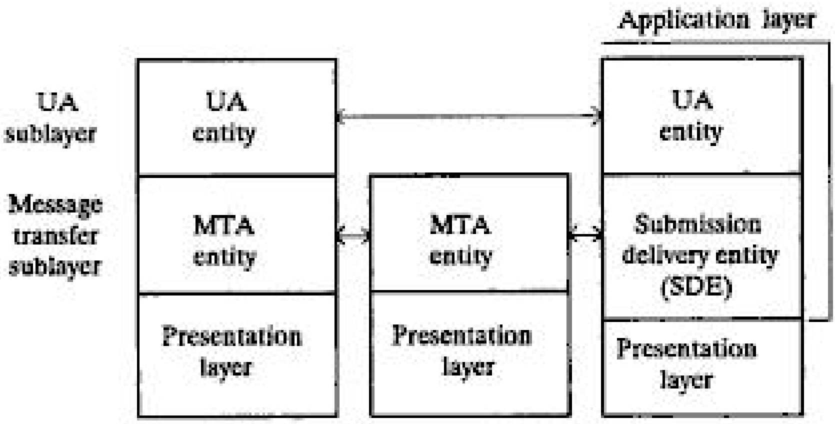

Fig. 11.4 X.400 in the context of OSI model.

plays a role and acts as a relay. Zero or more intermediate systems may be involved in transferring a message from the source MTAE to the destination MTAE. A message may be destined to one or more UAs and it is the responsibility of the message transfer sublayer to deliver the message to all the intended recipients. The MTAE may be asked to notify the source User Agent Entity (UAE) about the delivery status. It may also be given specific instructions about delivery time, etc. The services provided by the UAEs are known as interpersonal messaging services. Two types of services are supported:

- Send/receive user message
- Send/receive status reports.

Corresponding to the two types of services, two types of protocol data units (PDUs) are used. The PDU structure for exchanging user messages is shown in Fig. 11.5. The message structure consists of three parts. The body of the message is the actual user information, which may be considered as the user data unit in the OSI parlance. To this is added the control unit or the

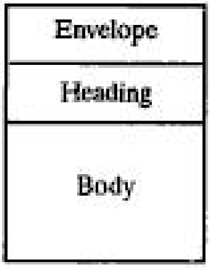

Fig. 11.5 X.400 user message format.

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S = frame size
I = information stream rate in bps
R = rate in bps to which the information stream is being adapted; 8, 16 or 32 kbps
D = number of information bits in each frame
R_F = frame rate in frames per second

The frame size and the frame rate must be so chosen as to satisfy the equations

$$I = R_F \times D, \qquad R = R_F \times S$$

For information streams with rates between 32 and 64 kbps, a single-stage rate adaption is used by padding unused bits with binary ones in a 64 kbps stream.

B channel may be used efficiently by multiplexing two or more low rate signals. Multiplexing is limited only to 8, 16 and 32 kbps streams. All other data rates must be first rate adapted to one of these rates and then multiplexed. Two schemes of multiplexing are suggested:

- Fixed format multiplexing
- Flexible format multiplexing.

The former is less complex than the latter and is generally suitable when the mixture of subrate streams is known in advance. In this scheme, an 8 kbps stream may use any of the eight bit positions in a B channel octet as information bit; a 16 kbps stream may use any of the four twin bits (1,2), (3,4), (5,6), or (7,8), and a 32 kbps stream may use one of the two quadruplets (1, 2, 3, 4) or (5, 6, 7, 8).

Fixed-format multiplexing may not lead to efficient B channel utilisation if the mixture of subrate streams vary dynamically. Consider a situation where three 16 kbps streams, 1, 2 and 3 are multiplexed in bits (1,2), (3,4) and (5,6), respectively of the octet. Let the data from stream 2 end and a new 32 kbps stream come up. The new stream cannot be accommodated in the octet under the fixed-format scheme, even though there are four vacant bit positions in the octet. Flexible format multiplexing scheme permits accommodation of this stream in the octet. In this scheme, a new subrate stream may be added by inserting each successive bit of the new stream in the earlier available vacant bit positions. It may be noted that under the flexible format scheme, subrate streams can be always multiplexed to the maximum limit of 64 kbps. We have so far been considering the use of B channel in circuit switched mode. ISDN permits the use of B channel in packet switched, semi-permanent and permanent modes as well. In packet switched mode, the data exchange is done by using X.25 protocol. Semi-permanent and permanent modes correspond to short-term and long-term connections. It must be remembered that when the B channel is used in modes other than packet switched mode, all multiplexed channels must be to the same destination.

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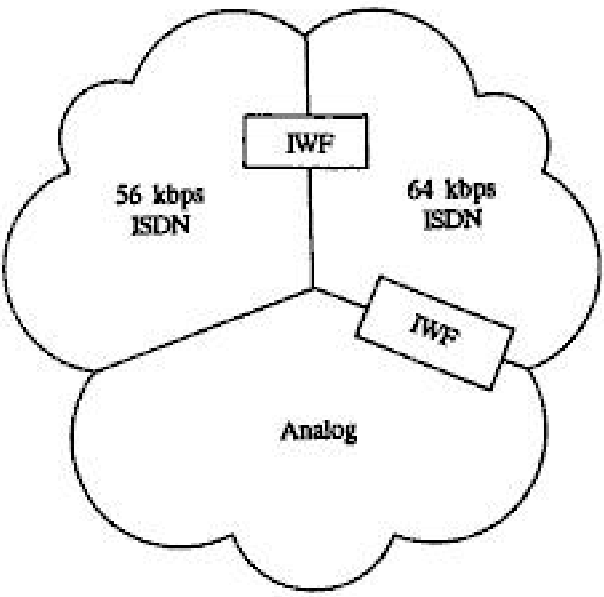
Low layer attributes are the same as the attributes for bearer services and in particular the information transfer and the access attributes. High layer attributes deal with different communication aspects such as voice, audio, text, facsimile and picture transmission. They are defined in the form of protocols conforming to the OSI model layers 4–7.

11.9 Interworking

ISDN is expected to evolve over a period of 20 years or so. During this period, it will have to coexist with other networks and this is essential for achieving market penetration and business success. Figure 11.18 depicts the situation.



IWF = Interworking functions

Fig. 11.18 Coexistence of ISDN with other networks.

At least three types of networks are involved: the analog networks, the 56 kbps digital networks for voice or data, and the ISDN with a 64 kbps clear channel. Interoperability is achieved through interworking functions (IWF). Interworking functions are different for different networks and so also the interface details. Taking this into account, separate reference points have been identified and their interface details have been defined to enable interworking with different networks:

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Index

Telecommunication Switching Systems and Networks

by
T. VISWANATHAN

Designed for the final year undergraduate or the first year postgraduate students in electronics and communication engineering and allied subjects, this compact and comprehensive text fulfils the long-felt need for a suitable text book in the area of telecommunication switching systems and networks. It covers topics of current interest such as fibre optic communication systems and networks, time division switching systems, data networks, ISDN and voice data integration schemes.

The book begins with a brief discussion on the evolution of telecommunication. It then goes on to give a classification scheme for switching systems, and describes the basic components of a switching system and the fundamental concepts of network structures. Principles of switching are illustrated through an analysis of electromechanical switching systems. Electronic switching and stored program control, digitalization of signals, and in particular speech signals, are also dealt with, followed by an analysis of digital time division switching. It provides an indepth coverage of the newly emerging communication system, viz. fibre optic communication system, and the traffic engineering concepts. A distinguishing feature of the book is the thorough treatment of the most important telecommunication networks, viz. public switched telephone network (PSTN), public data network (PDN), and integrated services digital network (ISDN).

This well-organized text lays emphasis on system-level treatment and practical aspects of telecommunication engineering. Worked out examples, as well as exercises and "Further Reading" given at the end of each chapter would be of considerable assistance to the reader. This comprehensive book, covering all aspects of telecommunication engineering, should prove to be an invaluable asset to both students and professionals.

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